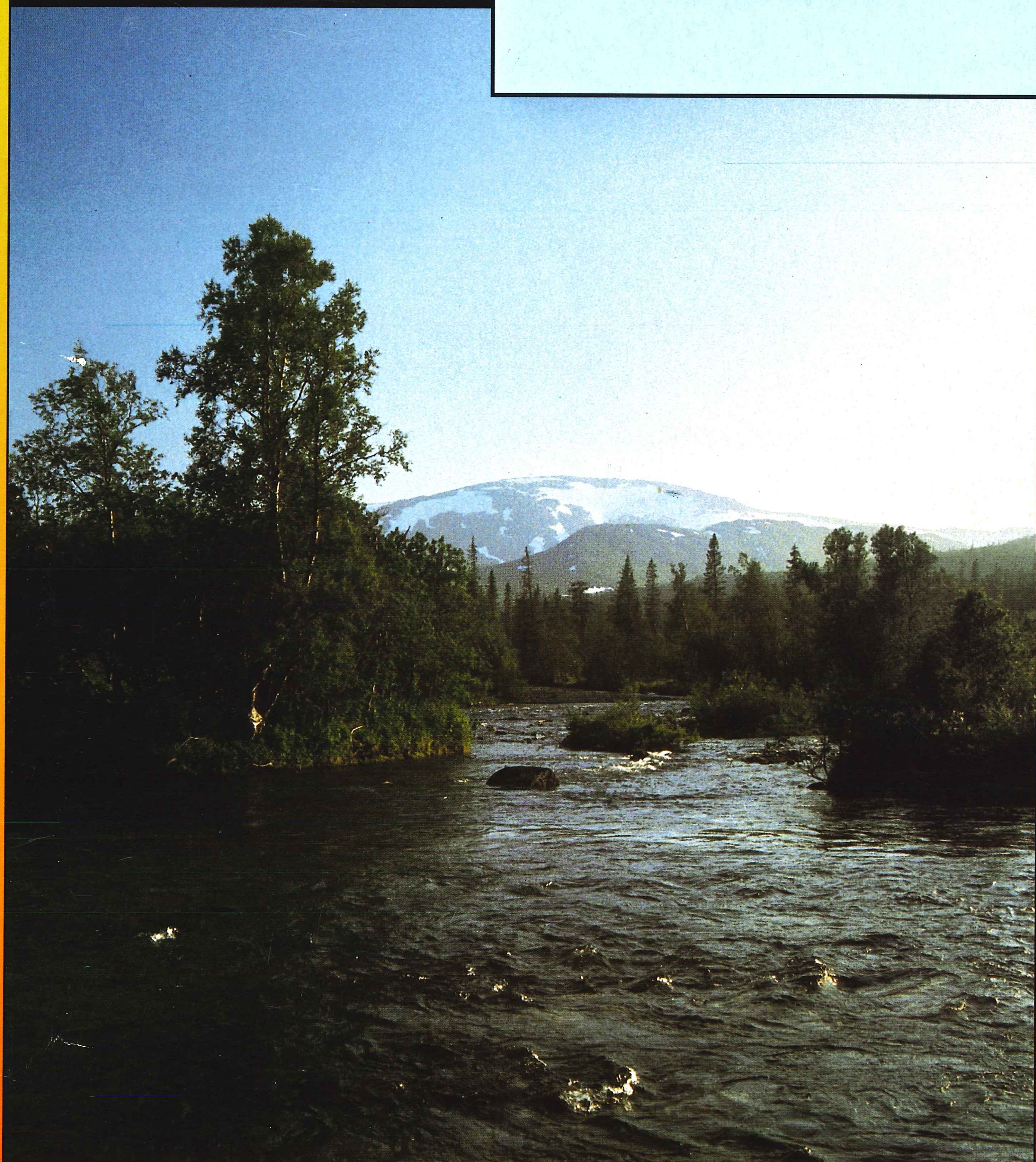
OKI ELECTRIC EUROPE GmbH

OKI

People to People Technology

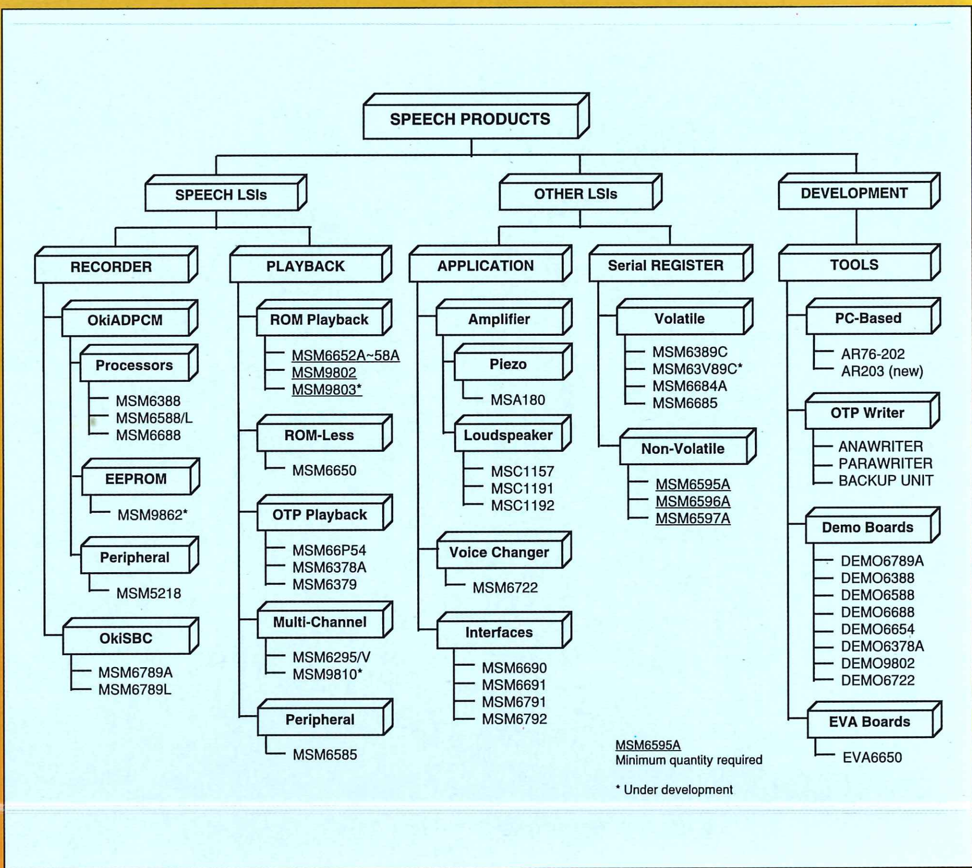
Speech LSI Products



November 1995

Introduction

OKI looks back on 114 years of history and experience in producing electric and electronic quality products of repute. OKI established its Electronic Devices Group in 1961, and has been designing and manufacturing high quality LSIs such as memories, microcomputers, telecom LSIs, a large variety of special purpose circuits and...OkiADPCM speech chips, ever since.

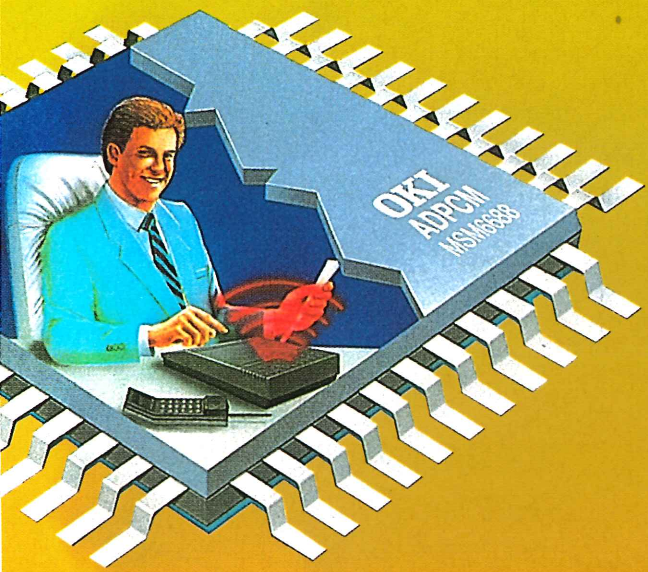


OKI´s speech LSIs enjoy remarkable popularity worldwide. Their superior speech quality, as a result of the company´s refined algorithm, has assured OKI a leading position among suppliers of speech chips.

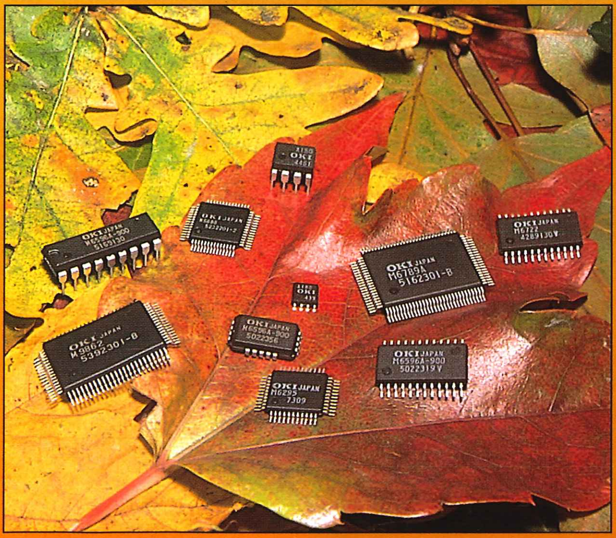
OKI speech circuits are easy to apply. Their internal circuit design allows a system based on an OKI speech chip to be operated as simply as a conventional tape recorder. Some circuits are designed for straightforward connection to a CPU or existing digital systems.

Speech LSIs are installed in many familiar products used in everyday life, such as answering machines, clocks, cameras, toys, telephones, greeting cards, office automation equipment, alarm systems, clinical facilities, musical instruments, etc. The range of applications is steadily expanding, and now also includes automotive use and the so-called brown and white goods. The facility of speech will in future be incorporated in an ever-increasing number of applications.

The new products introduced herein continue the tradition of the present range by placing unprecedented performance in the hands of the designer at an affordable price. OKI intends to continue to enhance a comprehensive product family aiming at ever increased performance and integration.



OKI´s OkiADPCM recorder devices are tailored for high demands in speech quality at equally high economy. With the new OkiADPCM voice processor MSM6688 and a Sub Band Coding processor MSM6789A this high speech quality can be applied for more than an hour recording time; this aspect is particularly interesting for telephone answering machines.



Chip form supply for selected products and small size packages save premium board space and allow full surface mount implementation thanks to OKI´s advanced IC packaging variety.

The Concept

OkiADPCM differs from other methods in that it does not truly synthesise speech. It initially involves digitising, compressing and storing actual analog sounds (digital recording). In order to re-create the original sound, the compressed digital word is expanded back to its original size, converted into an analog signal, amplified, and played through a speaker.

In fact, any kind of sound...voice, music, sound effects... etc., can be digitised, stored, and re-created faithfully with a high degree of naturalness. The idea behind this concept is to reduce the effective data rate in order to use memory economically and to minimise the data amount when transmitting voice data, while maintaining sufficient redundancy for faithful reproduction accuracy. In other words, maximum intelligibility and naturalness for all applications.

Common Features

- OkiADPCM, OkiSBC and OkiPCM algorithms,
- Suitable for speech and sound effects,
- High speech intelligibility and sound naturalness,
- Wide range of sampling frequencies,
- Full telephone bandwidth and more,
- Comprehensive functionality, extra functions,
- Easy to apply with only a few external parts,
- High design flexibility,
- Many analog extras on-chip,
- Reasonably priced,
- High device quality and reliability,
- Through-hole & SMD packages,
- Low power CMOS process technology.
- Typical applications:
  Handicap aids, Medical systems, Teaching aids, Answering machines, Telecommunication, Industrial equipment, Computer systems, Handies, Consumer goods, Toys...and more.

Application circuits contained herein are designed to give the reader merely an idea about typical applications and the components required. They do not represent application notes.

OkiADPCM Recorders

MSM5218 - Recording For Prototypes

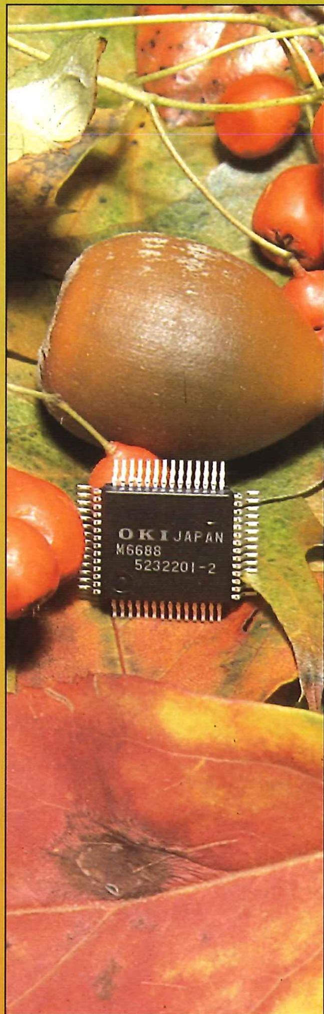
This LSI enjoys world-wide popularity thanks to its real-time analysis/synthesis capability and high versatility in use. The inexpensive MSM5218 gives the user nearly unlimited design flexibility. External conversion accuracies from 8 to 12 bits and a wide range of adjustable sampling frequencies allow economical designs with optimised voice quality. A built-in data overflow protection circuit minimises distortions as a result of excessive analog inputs during recording. MSM5218 can be added to any existing digital system.

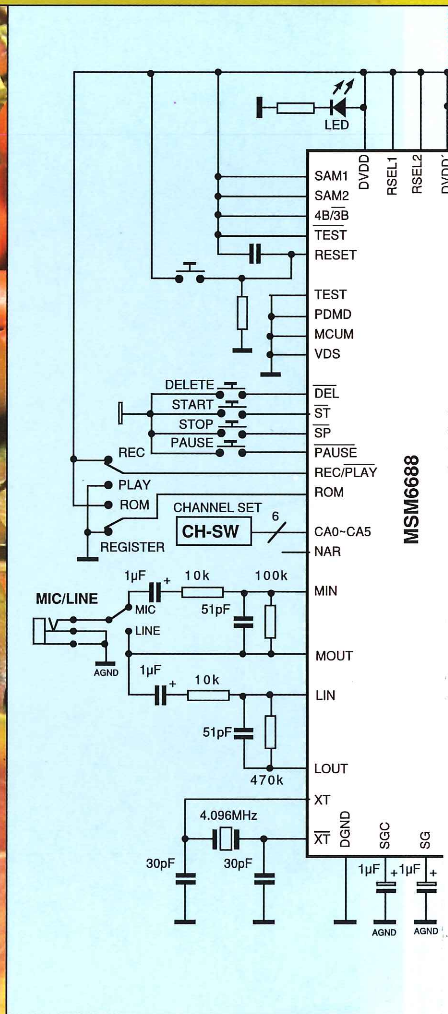
MSM6585 - Playback For Prototypes

Functionally identical to MSM5205 (an earlier best-selling synthesiser with 10-Bit DAC), this device comes with a 12-bit D/A converter plus internal low-pass filter. Apart from saving parts count, MSM6585 offers superior reproduction fidelity due to the higher accuracy of its D/A converter. To allow easy upgrading of existing MSM5205 boards with MSM6585, the pin assignment has been largely maintained. Provided a MSM5205 board is operated at 4-bit ADPCM data, the devices can be exchanged in the socket with only the clock resonator replaced. Owing to the high reproduction accuracy, MSM6585 is recommended for CD-ROM applications, such as car navigation equipment.

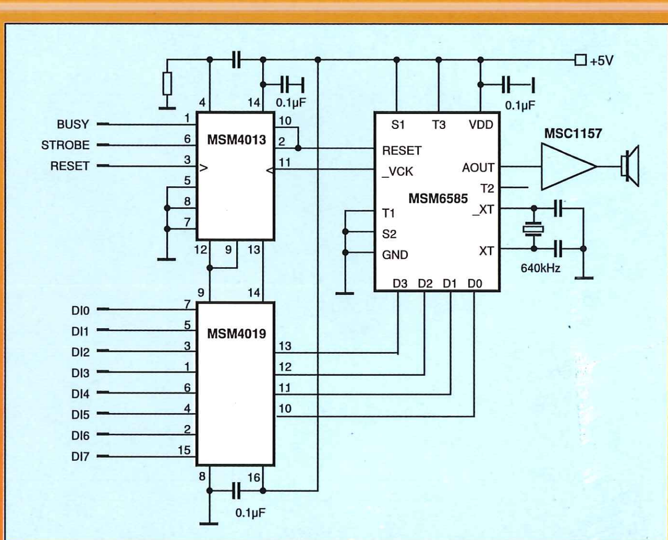
MSM6388 - A Bestseller

The main attractions are undoubtedly the internal 12-bit ADC and 12-bit DAC which guarantee a significant gain in speech quality. The maximum I/O time of 262 seconds at 4KHz sampling is achieved by means of an interface that controls four voice register LSIs, such as the OKI MSM6389B. MSM6388 is pin-switchable between stand-alone and MCU interface modes. Digital and analog circuits are combined on the same chip, including two OpAmps and a low-pass filter. The master clock is 4.096 MHz typically allowing a sampling range from 3.5 to 9.1 KHz. The sampling frequencies can be increased proportionally if a higher clock frequency is selected. The available memory capacity can be split into 1 to 8 channels in the stand-alone mode. Operation control in the MCU mode allows more than 8 channels and is implemented over a 4-bit control bus, by which the user accesses numerous registers to control start, stop, record, play, start-stop addresses, sampling frequencies, etc. In total, 16 powerful instructions are built in for high application convenience. MSM6388 is particularly suitable for full or half-solid state answering machines with up to 4 minutes recording time.

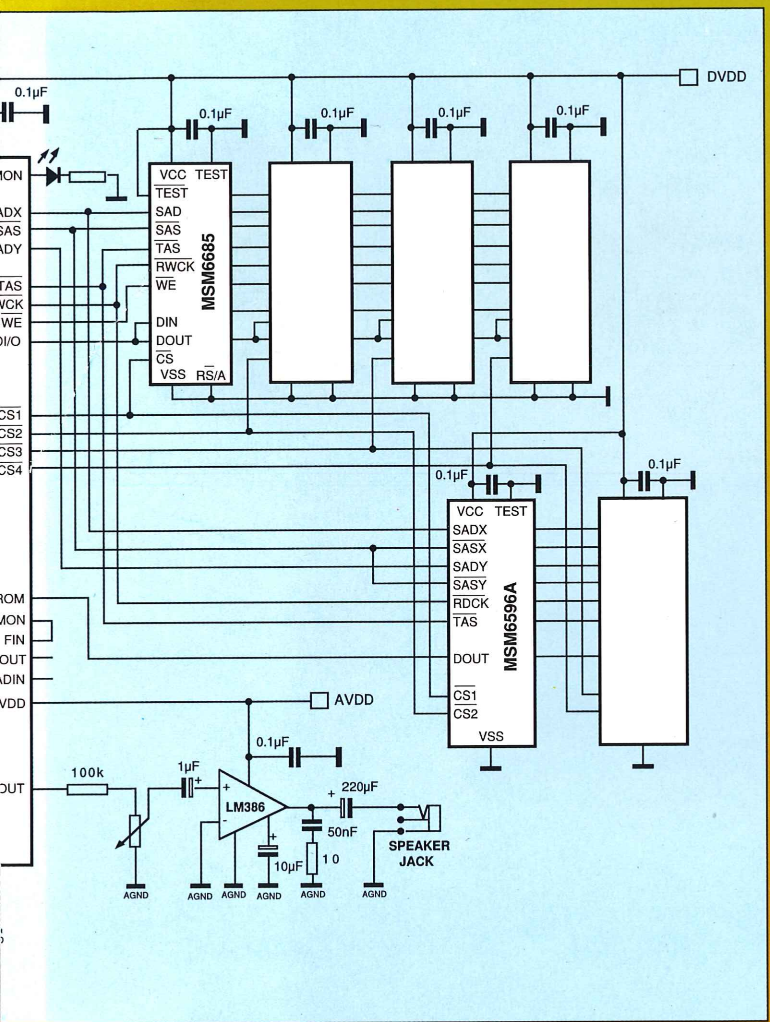




TYPICAL APPLICATION WITH MSM6688 IN CONJUNCTION WITH 8-MEGAB



SIMPLE CENTRONICS INTERFACE WITH MSM6585



SERIAL VOICE REGISTERS AND ROM .

2 MEGABIT SERIAL VOICE ROMs IN THREE PACKAGES

MSM6588 - Another Bestseller

With basically the same features as MSM6388, the MSM6588 additionally provides voice triggered starting of the record function, pause mode and a simplified, but equally powerful set of 13 commands with 4 times faster processing speed in the MCU mode. Another extra for the stand-alone mode is the possibility to define 1 to 8 recording channels with fixed or flexible length. A hardware reset input has also been included. MSM6588 operates at 3-bit ADPCM in the stand-alone mode and at 3 or 4 bit selectable in the MCU mode. Shortly, also available as 3V version, the MSM6588L.

MSM6688 - For Long Recording Times

MSM6688 is essentially based on MSM6588, but incorporates additional features designed for long-time answering machines. It addresses up to 32-Megabit memory resulting in a considerable recording time of 35 minutes at 16kbps. Like MSM6388 and MSM6588, it provides both MCU and stand-alone operating modes, both supporting 3-bit or 4-bit ADPCM data. Its stand-alone mode provides 63 recording channels administered by a data header on top of the interfaced register bank. In the MCU mode, a powerful instruction set of 14 commands makes full use of the speech processor´s capabilities, including a preset of recording time length. Voice trigger, channel erase, various recording modes, speech/address data transfers and a hardware reset are useful extras in both modes. Dynamic RAM is used in conjunction with an interface circuit, MSM6791, for incoming and outgoing messages. If MSM6688 is used as a peripheral chip the interface LSI is dispensable. Thanks to a separate data input, up to 4Mbit voice ROM can be added to the maximum memory capacity of 32Mbit without compromising address space.

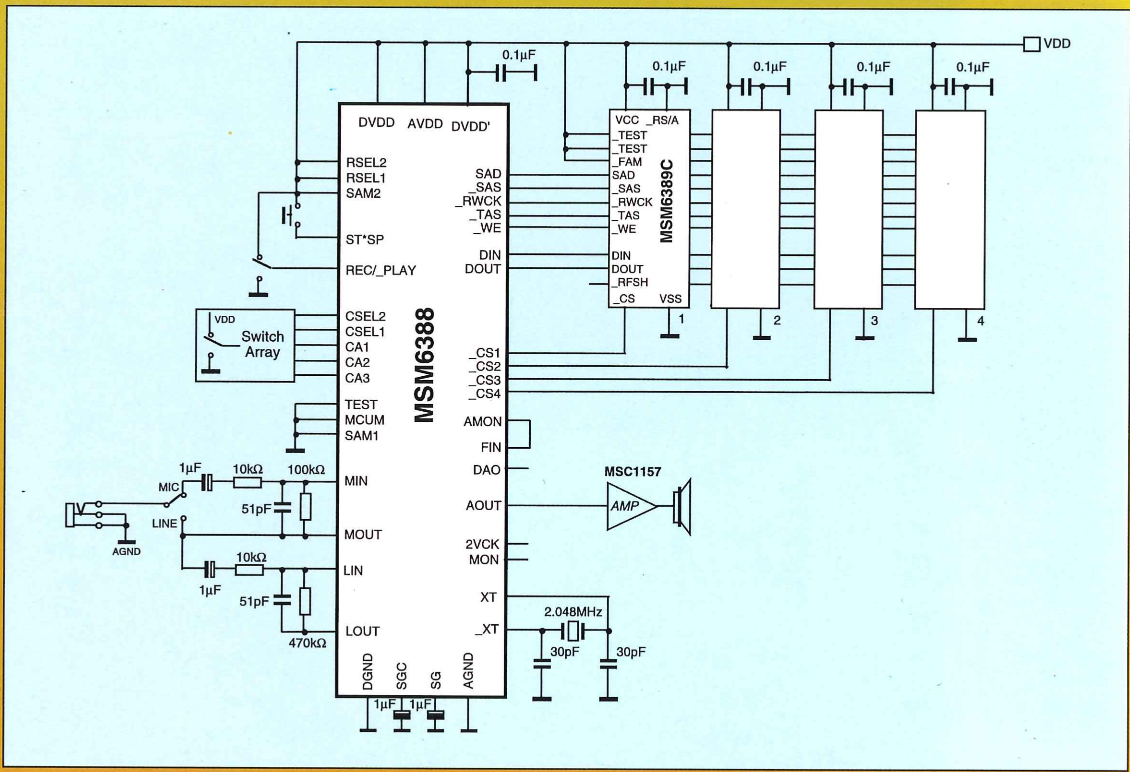
Economical Voice Registers and Voice ROMs

Serial registers are designed to interface with speech processors to store compressed speech data. Four registers are serially addressed by the voice LSIs. The capacity for recording and playback can be expanded significantly by switching multiples of four voice registers. The register access cycle time is 2.5µs to 4µs while the current consumption is as low as to allow battery backup for extended periods. 1-Megabit registers are designed for MSM6388 and MSM6588/L. A 3V version of the 1 Megabit serial register, MSM63V89C will be introduced, shortly. Designed for the speech processors MSM6688 and MSM6789A/L are 4 and 8-Megabit serial registers, representing a true alternative to conventional dynamic RAMs

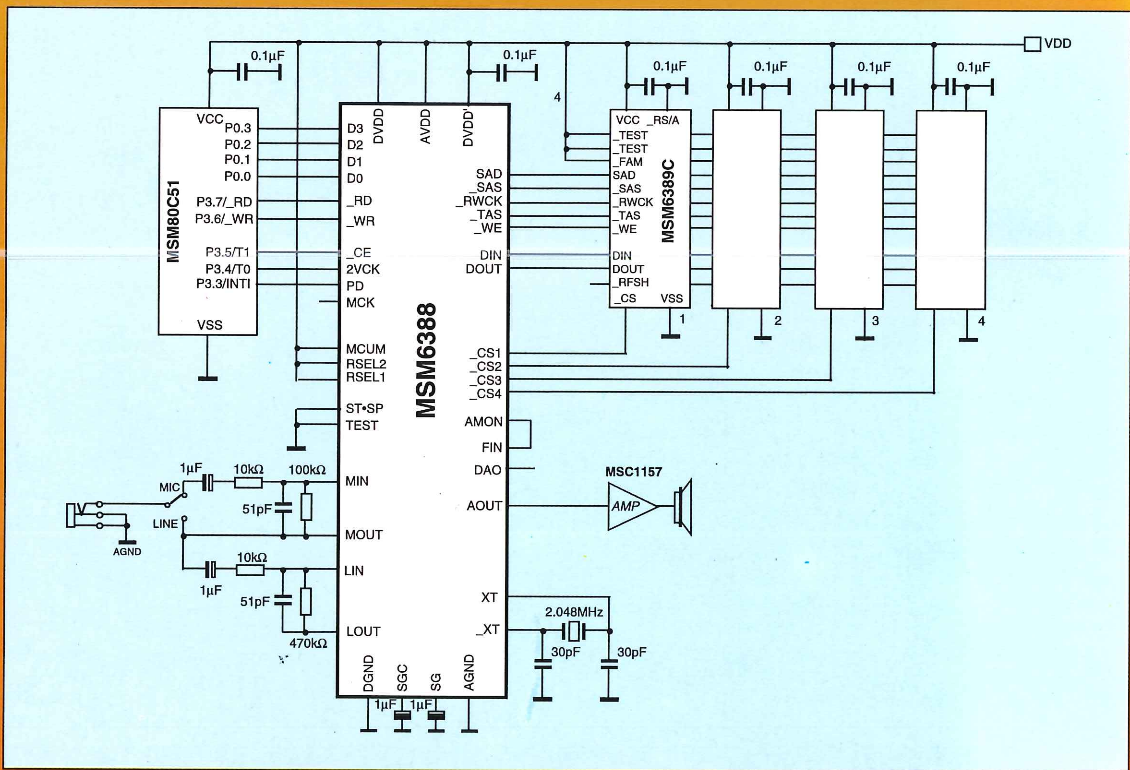
Serial voice ROMs are used in conjunction with serial registers for the storage of non-volatile outgoing messages such as time stamps or voice prompts. They are available in SMD packages with 1, 2 or 3-Megabit capacities per chip, while the latter two are internally split into one Megabit banks switched by separate select inputs.

Serial voice registers and serial voice ROMs combine the advantages of DRAMs (high capacity) and SRAMs (low current) in smaller packages.

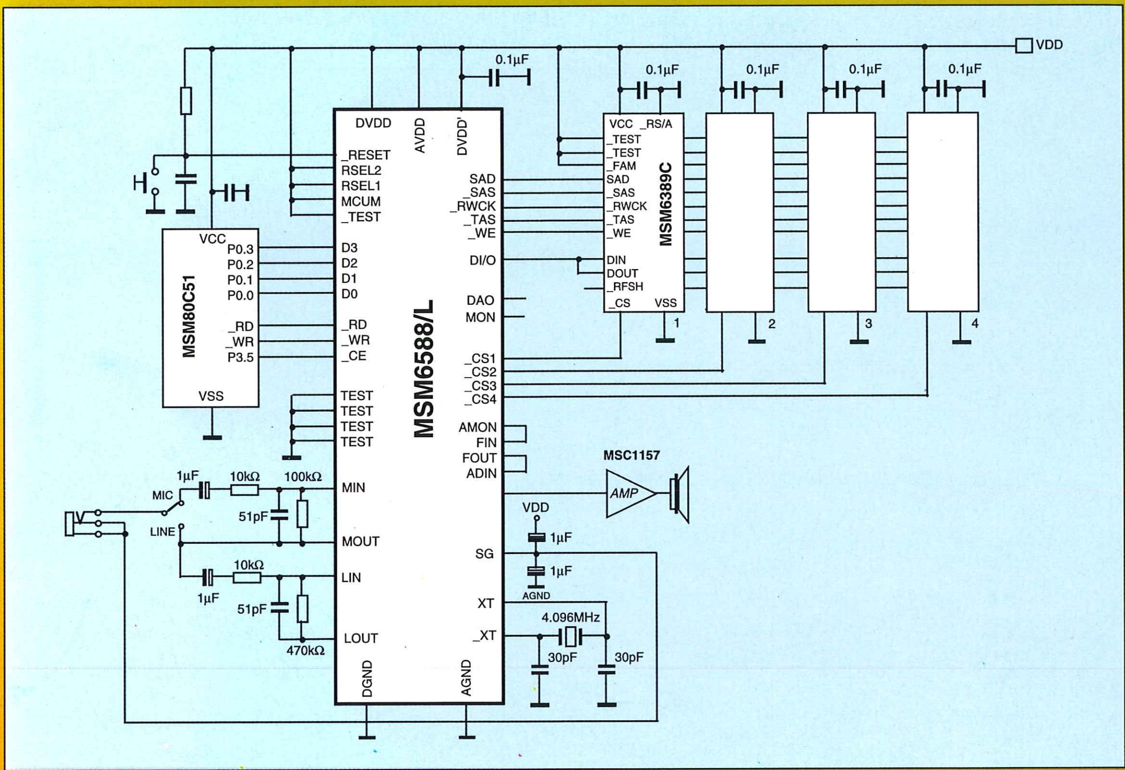
OkiADPCM Recorders



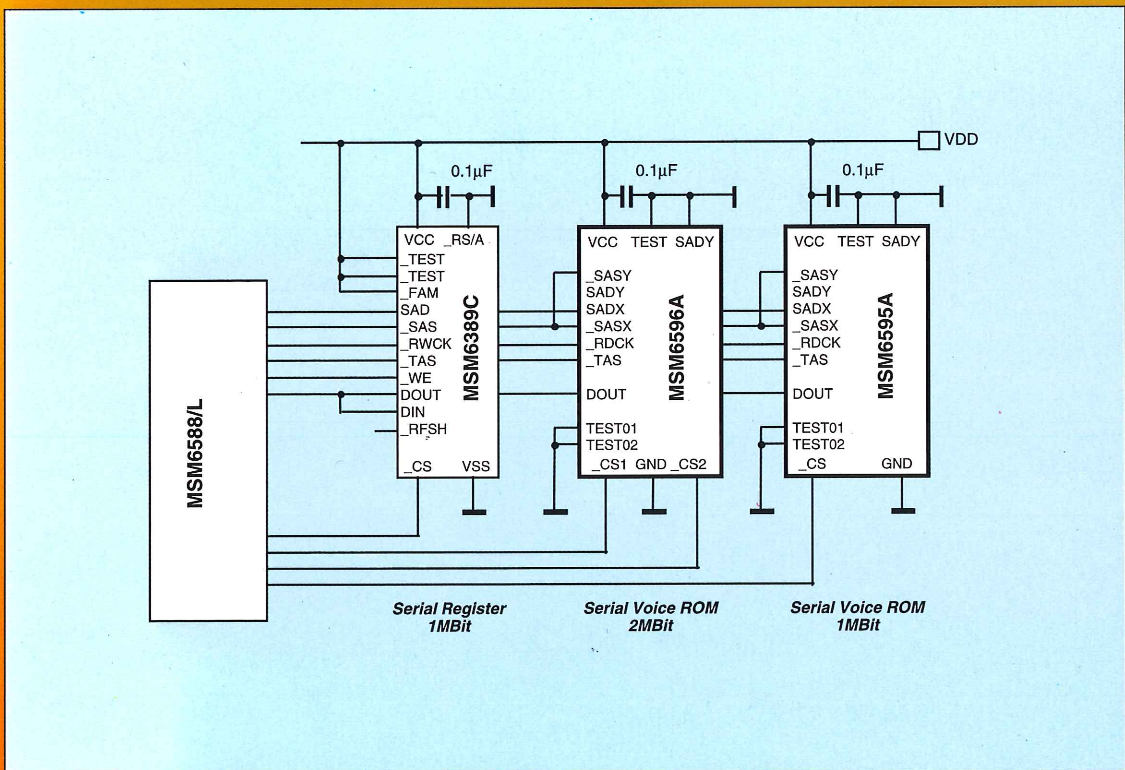
TYPICAL STAND-ALONE APPLICATION WITH MSM6388



TYPICAL CONTROLLER APPLICATION WITH MSM6388



TYPICAL CONTROLLER APPLICATION WITH MSM6588/MSM6588L



TYPICAL SERIAL VOICE ROM INTERFACE, SHOWN WITH AN MSM6588/MSM6588L

OkiSBC Recorder

MSM6789A: Low Bit-Rates With OkiSBC

Traditionally, OKI Electric uses the ADPCM method to facilitate data reduction. Our Company is now introducing a speech processor LSI which implements higher compression rates. To achieve this, the Sub-Band-Coding method, OkiSBC, has been chosen in the interest of both longer recording time and good reproduction quality. Nevertheless, the functional advantages of the OkiADPCM processors are maintained.

The OkiSBC processor is of particular interest to the telephone answering machine industry demanding 30 minutes or more recording time at economical cost for a solid state solution. OkiSBC is a true alternative if quality is not the feature to trade off. With MSM6789A, the user will be given the choice of various bit-rates, 7.5, 9, 10, 12, 12.6 and 16Kbps typically with programmable silence deletion function. Naturally, the speech reproduction fidelity lowers as the bit-rate is decreased. Supposing 10Kbps selected, the maximum recording time in conjunction with 16-Megabit memory is approximately 28 minutes. With the maximum capacity of 32 Megabits, considerable 56 minutes are within reach. At other bit-rates and memory size, recording times change proportionally.

In its stand-alone mode, MSM6789A provides 63 recording channels with fixed or flexible length. Under the control of an MCU, such as an OKI MSM83C154, all essential functions are addressed and executed by means of a powerful instruction set. In total 63 recording channels are selectable by register addressing. More channels can be defined if the MCU issues start and stop addresses from an internal look-up table. Also implemented is voice triggered recording with programmable threshold levels. Built-in analog functions (12-bit converters, OpAmps and low-pass filter) minimise parts-count for external components allowing for direct connection of a microphone and power amplifier.

The interface to memory is straight-forward eliminating the need of line drivers or buffers. It supports up to 32 Megabit capacity in various configurations. Selectable DRAM refresh cycles permit the MSM6789A to save premium power. In addition to volatile memory, up to 4-Megabit non-volatile serial voice ROMs can be connected for the purpose of installing fixed outgoing messages, such as time stamps or voice prompts. This addition does not compromise address space for recordings and also allows playback of linear PCM data or non-linear OkiPCM data, while up to 256 channels are selectable. Speech codes for serial voice ROMs are generated by means of OKI´s PC-based development system. If the MSM6789A is used as a peripheral chip without memory addressing, the MCU provides memory control and passes speech and command data through the MSM6789A.

MSM6789A is the truely superior choice for designs which are to combine long recording times with high intelligibility and operating convenience. A 3V version of MSM6789A, the MSM6789L will be introduced, shortly.

Principle of Sub Band Coding, SBC

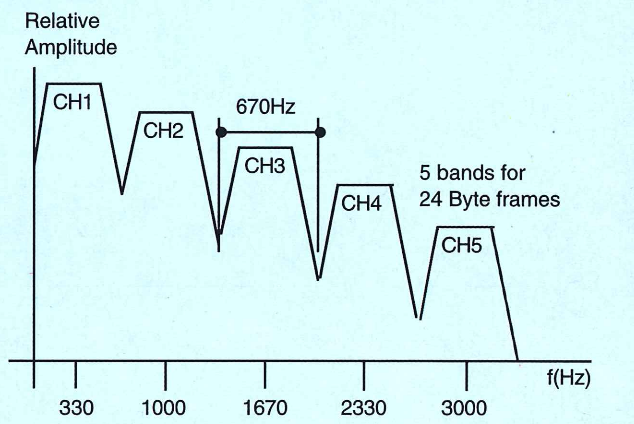
This algorithm involves the splitting of a digitised input signal into typically five frequency bands using a filter array. The centre frequencies of each band are 330Hz, 1.0kHz, 1.67kHz, 2.33kHz and 3.3kHz, each with a bandwidth of 670Hz. Each sub band signal is then quantised and stored by means of a block companded PCM quantiser, BCPCM, which features a wide dynamic range. The BCPCM block detects the maximum amplitude of each block to quantise with reference to a logarithmically scaled table in ROM which also determines the step size of the quantiser. As a result, the sign of a sub band and its average amplitude value is stored in memory. The decoder recovers the signal in the previously generated bands. It is then filtered, summed and provided as the output signal.

OkiSBC stores speech data by means of frames each containing 96 samples. The number of bytes being stored for one frame depends on the bit-rate, 24 at 16 kbps, 19 at 12.6kbps and 15 at 10kbps. The available bit-rates vary with the sampling frequency, 6 or 8kHz. In case of 24-Byte-frames the analog waveform is split into 5 bands, otherwise in 4 bands. Independent from the bit-rate, the time that one frame reproduces only depends on the sampling frequency, 12ms at 8kHz and 16ms at 6kHz.

As samples are stored frame by frame and because the algorithm, unlike OkiADPCM, does not make reference to previously computed values, the occurance of bit-failures will have only little impact, which can hardly be perceived. However, any header area in the speech memory must be error free. The OKI MSM6789A speech processor uses this algorithm to achieve bit-rates as low as 7.5 to 16kbps, plus silence level detection and removal. The typical cut-off frequency of the internal low-pass filter is 0.4 times the sampling frequency.

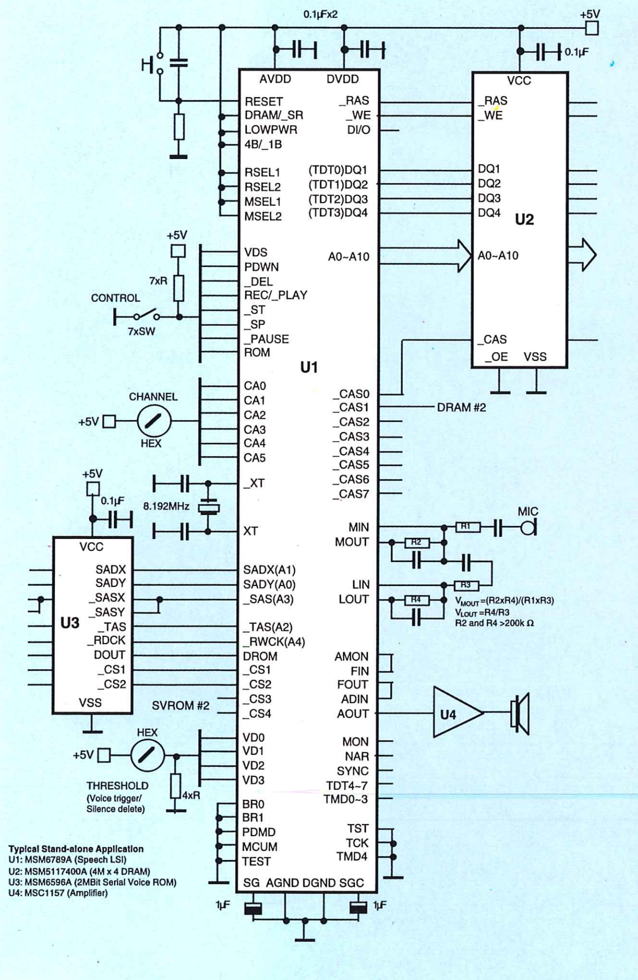
fSAMPLE	Bit-rate	Bytes	tFRAME	Samples
6KHz	7.5Kbps	15	16ms	96
	9.5Kbps	19	16ms	96
	12.0Kbps	24	16ms	96
8KHz	10.0Kbps	15	12ms	96
	12.6Kbps	19	12ms	96
	16.0Kbps	24	12ms	96

$$\textit{Bit-rate} = (\#\textit{Bytes} \times 8) \,/\, t_{FRAME}$$

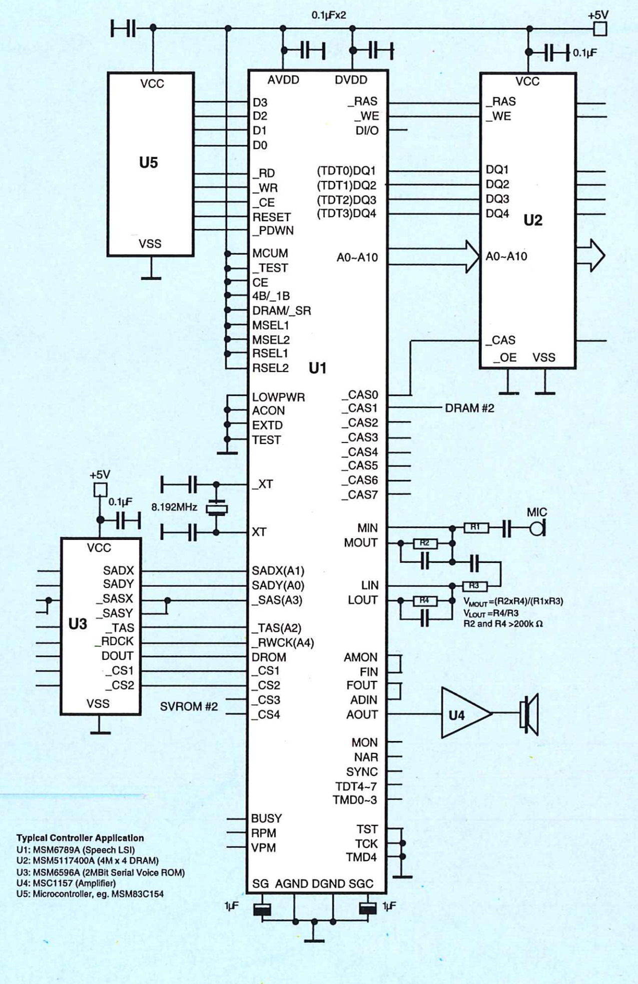


OkiSBC FRAME CODING AND TYPICAL BAND SPLITTING

Typical Application Diagrams



TYPICAL STAND-ALONE APPLICATION

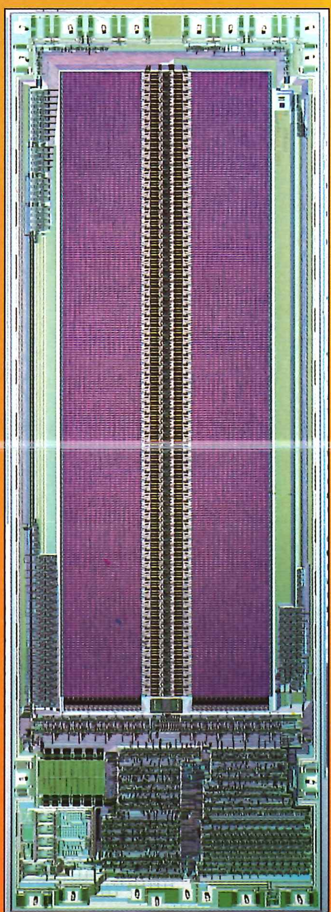


TYPICAL MCU CONTROLLER APPLICATION

Speech Synthesiser

MSM6650 Family - Mask ROM Synthesisers

HIGH CAPACITY MASK ROM SYNTHESISERS



MSM6658A CHIP, 4MBIT ROM

The MSM6650 family consists of six ROM versions, one OTP version and one evaluation version for external memory. They are based on OKI Electric's successful MSM6375 family and feature upgraded functions and performance. Depending on mask selection, the LSIs operate on a stand-alone basis or controlled by an MCU. In the stand-alone mode, speech is reproduced immediately upon address selection with no more start pulse required. In the MCU mode, two basic means of control are provided, a parallel bus or a serial bus, either of which is selected by mask option. If the serial interface is selected, two MCU I/Os are added by the synthesiser. The quality of the speech reproduction is based on the sampling frequency, which can be selected from a range of 4 KHz to 32KHz. Alternatively, linear 8-bit PCM coding can be used. Moreover, each phrase may be sampled at different sampling frequencies or different data formats in order to achieve maximum memory economy whilst obtaining optimum speech quality. Besides the ability to form sentences from 127 different phrases, the following additional features have been implemented:

- Melody generation,
- Two-channel or echo reproduction,
- Level attenuation,
- Fade out function,
- Four different beep tone frequencies,
- Random playback of phrases.

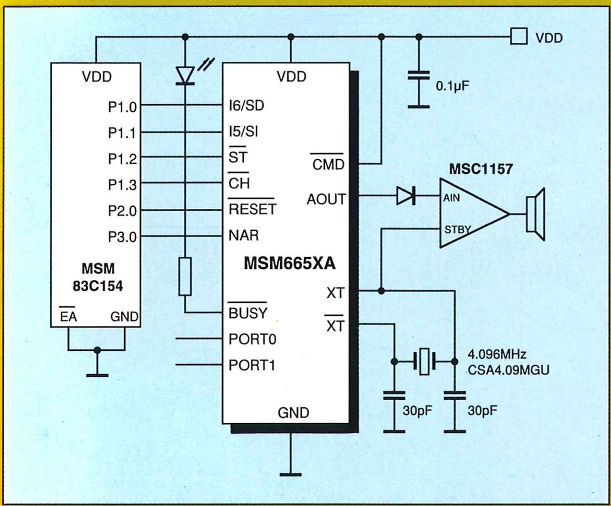
The melody function is realised by means of a compression that takes advantage of data repititions and results in an average bit-rate of 4 to 5 kbps. Maximum voice quality is achieved by the internal 12-bit D/A converter and 4th order low-pass filter. What is more, the LSIs come in low pin-count packages or in chip form and offer a new approach to typical applications such as telephones, facsimile equipment, household appliances, alarm systems, AV, computers, measuring equipment, watches, thermometers and, of course, games and toys. Voice codes for masking are easily generated by means of OKI´s PC based tool and directly placed in an EPROM. For MSM6650 (ROM-less version), the voice EPROM can be used directly in the application circuit, while up to a maximum of 64Mbit external memory can be addressed. The OTP-version is available with 1 Megabit capacity for prototyping or smaller volumes.

Product Line-up

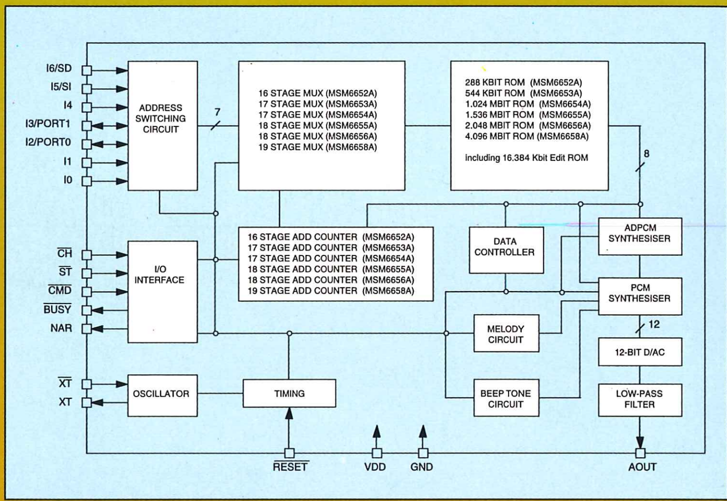
Part Number	Package	ROM Size	Max Time
MSM6652A	18-DIP/24-SOP	288Kbit	16.9 sec
MSM6653A	18-DIP/24-SOP	544Kbit	33.2 sec
MSM6654A	18-DIP/24-SOP	1.024Mbit	63.8 sec
MSM6655A	18-DIP/24-SOP	1.536Mbit	96.5 sec
MSM6656A	18-DIP/24-SOP	2.048Mbit	129.1 sec
MSM6658A	24-SOP	4.096Mbit	260.0 sec
MSM6650	64-QFP/64SDIP	ext. 64Mbit	70 min
MSM66P54	20-DIP/24-SOP	1.024Mbit	63.8 sec

Maximum playback times are based on ADPCM at 4 kHz sampling frequency (ADPCM bit-rate 16kbps).

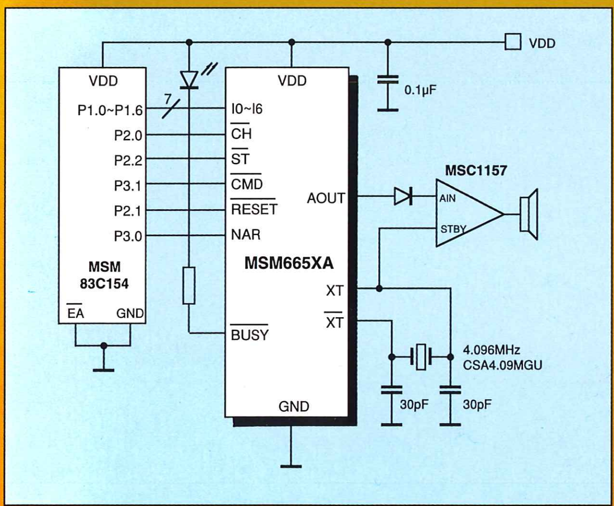
MSM6650 Family



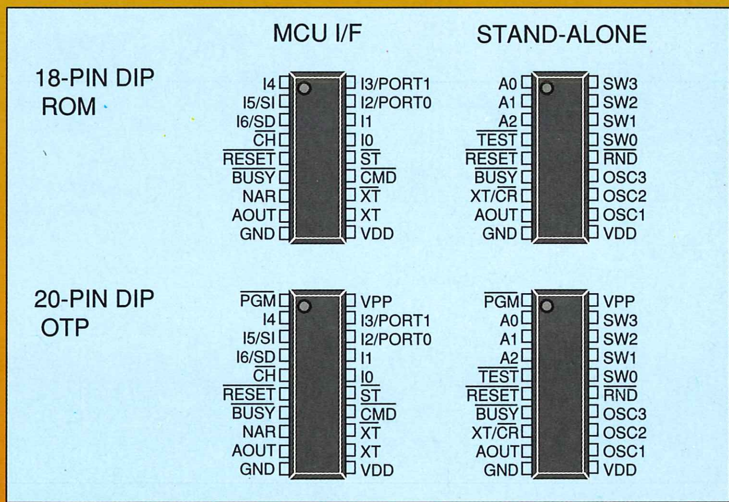
TYPICAL SERIAL CONTROLLER MODE



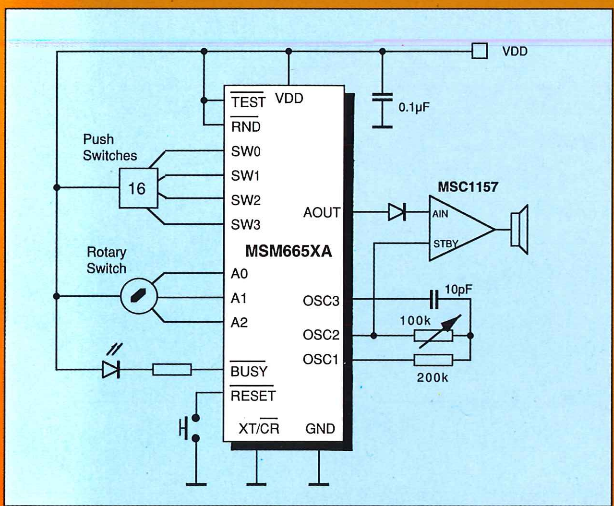
BLOCK DIAGRAM MCU MODE



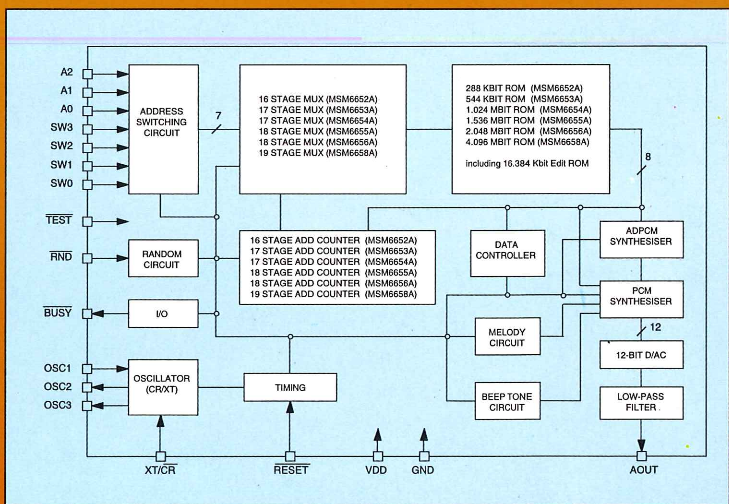
TYPICAL PARALLEL CONTROLLER MODE



PIN ASSIGNMENTS ROM vs. OTP VERSION (DIP)



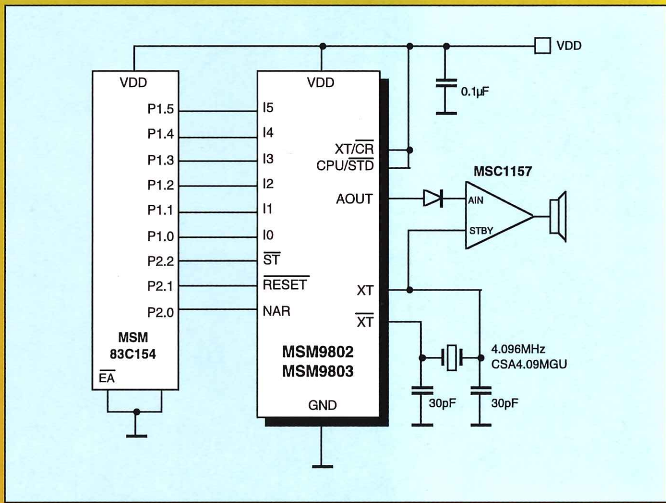
TYPICAL APPLICATION WITH STAND-ALONE



BLOCK DIAGRAM STAND-ALONE MODE

MSM9802/MSM9803

OkiPCM / LINEAR PCM SPEECH SYNTHESISER MSM9802



TYPICAL CONTROLLER APPLICATION

MSM9802/MSM9803 - PCM Synthesisers

These speech synthesisers are designed for astounding speech quality for relatively short output times and differ from other OKI synthesiser products in that they do not use OkiADPCM but PCM reproduction. Two methods are selectable, linear 8-Bit PCM or non-linear 8-Bit OkiPCM. The latter is a refined algorithm and provides a quality comparable with 10-Bit linear PCM.

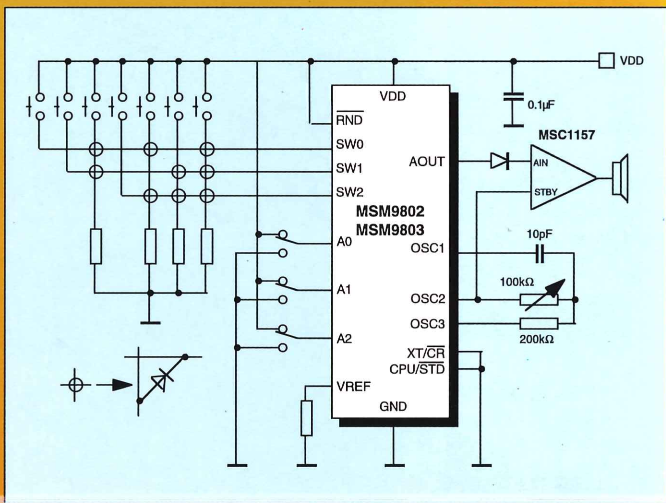
MSM9802 comes with built-in 512KBit ROM, MSM9803 with 1MBit and address consumer and professional applications requiring high speech and sound fidelity as much as simplicity in use. In fact, only a few external components and a power amplifier (or transistor) are required to establish a high quality speech playback unit. Both manual switch and MCU control modes are available by means of mask option. Integrated on-chip is an edit ROM function by which means it is possible to define complete sentences for playback by applying only a single address.

A wide choice of selectable sampling frequencies from 4 to 16KHz satisfy all demands in speech quality. The maximum playback time is achieved by MSM9803 with 32 seconds at 4KHz sampling. Moreover, each phrase may be sampled at different sampling frequencies or different data formats in order to achieve maximum memory economy whilst obtaining optimum quality for specific phrases containing speech or sound effects.

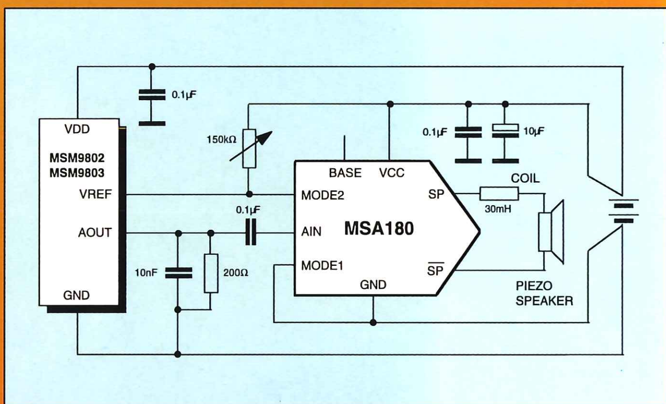
The analog chip portion contains a 10-bit DAC and a low-pass filter.

Operation modes include MCU or switch interfaces, 63 phrase channels and random playback (in stand-alone only). Also provided is an Edit ROM area for the precompilation of phrases, as similarly performed by the MSM6650 family.

The chips can be clocked using either a RC network at 256KHz or a crystal at 4.096MHz, while the MCU control mode requires the crystal.



TYPICAL STAND-ALONE APPLICATION



MSM9802/9803 USING THE PIEZO DRIVER MSA180

MSM6378A/MSM6379

MSM6378A/MSM6379 - Programmable OTPs

MSM6378A is a voice synthesiser including 256Kbit OTP (one-time programmable ROM), while MSM6379 is the equivalent with 512Kbit ROM. Internally comprising a 12-bit DAC and a buffered low-pass filter, the devices merely require an external amplifier, a speaker and an RC network to be fully operable. For programming of the internal OTP, OKI offers the ANAWRITER which is a tool designed for recording an analog source (without editing), conversion to OkiADPCM data and device programming. Alternatively, the PC-based tool can be used with all editing possibilities.

The vocabulary is stored in the OTP as one single phrase, which can be reproduced once or by means of endless-loop playback. The sampling frequencies are derived from the oscillation frequency of 64 to 256KHz through division by 16. Consequently, sampling frequencies from 4 to 16KHz are practicable, resulting in a maximum playback time of up to 32 seconds with MSM6379 at 4 KHz sampling. Thanks to the one chip solution, MSM6378A and MSM6379 suit a wide range of applications, including personal voice card and other miniature devices designed to produce high quality speech.

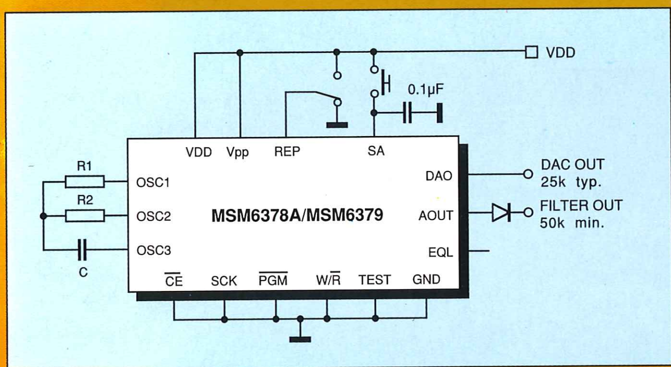
MSM6378A AND MSM6379 ARE OFFERED IN 16-PIN DIP

MSM6378A/MSM6379 Programmer´s Kit

The ANAWRITER is a dedicated tool which records an analog source through a microphone or from a tape into its buffer memory allowing a verify playback of the voice code thus recorded. The target device MSM6378A or MSM6379 is inserted in a Textool™ socket and programmed within 8 and 16 seconds, respectively.

More development convenience is offered by two additional units, namely the BACK-UP WRITER and the PARAWRITER. The BACK-UP WRITER is connected to the ANAWRITER and allows a verified voice code to be transferred from the ANAWRITER into an EPROM. This master EPROM is then used as the source from which the PARAWRITER programs 10 devices at a time requiring only a few seconds more than the progrmming of a single device. Such master EPROMs can also be witten using the PC-based tool.

With these tools, the user enjoys maximum flexibility by developing OTP synthesiser codes in-house for small, medium and larger quantities.



ONE-TIME PROGRAMMABLE: MSM6378A/MSM6379

´ANAWRITER´ PROGRAMMING SYSTEM

Specification Overview

Speech Recorder LSIs

PARAMETER	MSM5218	MSM6388	MSM6588/L	MSM6688	MSM6789A/L	MSM9862
Algorithm	OkiADPCM	OkiADPCM	OkiADPCM	OkiADPCM	OkiSBC	OkiADPCM
Sample Data Length	3/4-bit	4-bit	3/4-bit	3/4-bit	-	3/4-bit
Osc. Frequency (Hz)	384~768K	1.5~8.192M	4~8.192M	4~8.192M	6~8.2M	4.096M
Bit-rate (bps)	16~64K	12.8~72.8	16~64K	16~64K	7.5~16K	4~8
External Memory	RAM/ROM	4M-Register	4M-Register	32M-DRAM	32M-DRAM	4M ROM/REG
Typ. I/O Time (min)	variable	4.3	4.3	35	74	0.5+ext.
Number of Channels	variable	1~8	1~8	1~63	1~63	1~8
Internal LPF	no	-40dB/oct	-40dB/oct	-40dB/oct	-40dB/oct	-40dB/oct
Internal A/D Converter	no	12-bit	12-bit	12-bit	12-bit	12-bit
Internal D/A Converter	10-bit	12-bit	12-bit	12-bit	12-bit	12-bit
Supply Voltage	+3~6V	+3.5~5.5V	+3.5~5.5V	+3.5~5.5V	+4.5~5.5V	+4.5~5.5V
Active Current (max.)	6mA	10mA	15mA	30mA	35mA	TBA
Operating Temperature	-30~+70°C	-40~+85°C	-40~+85°C	-30~+70°C	0~+70°C	-40~+85°C
Packaging	24-DIP	44-QFP	44-QFP	56S-QFP	100-QFP	80-QFP
Notes	As voice peripherals to hosts	Stand-alone & MCU control	Stand-alone & MCU control, Voice trigger. L-version: VDD=3.0~3.6V	Stand-alone & MCU control. MSM6791 as DRAM I/F required.	Stand-alone & MCU control. L-version: VDD=3.0~3.6V	DTMF Rx, 512K Flash E²PROM, ext. ser ROM, ext. ser REG,

• Registers refer to serially addressable voice registers, such as MSM6389C (volatile) and MSM6595A, MSM6596A, MSM6597A (non-volatile).
• Typical output times are given for minimum the bit-rate and with the chip´s own memory addressing capability without expansions. Number of channels apply for the stand-alone mode, and is flexible if operated in the MCU mode.

Speech Playback LSIs

PARAMETER	MSM6295	MSM6378A	MSM6379	MSM6585	MSM9802	MSM9803	MSM9810
Algorithm	OkiADPCM	OkiADPCM	OkiADPCM	OkiADPCM	PCM/OkiPCM	PCM/OkiPCM	PCM/OkiADPCM2
Sample Data Length	4-bit	4-bit	4-bit	4-bit	8-bit	8-bit	8bit/4-bit
Osc. Frequency (Hz)	1~5M	64~256K	64~256K	640K	256K/4.096M	256K/4.096M	4.096M
Sampling Rate (KHz)	6.5~32	4~16	4~16	4~32	4~16	4~16	4~32
External Memory (bit)	2M ROM	-	-	-	-	-	128M
Internal Memory (bit)	-	256K OTP	512K OTP	-	512K ROM	1M ROM	-
Typ. Output Time (sec)	78	15.6	32	variable	16	32	134 min
Number of Channels	127	1	1	variable	63	63	256
Internal LPF	no	-24dB/oct	-24dB/oct	-40dB/oct	yes	yes	yes
Internal D/A Converter	12-bit	12-bit	12-bit	12-bit	10-bit	10-bit	14-bit
Supply Voltage	+4.5~5.5V	+2.7~5.5V	+2.7~5.5V	+4.5~5.5V	+2.4~5.5V	+ 2.4~5.5V	+ 2.7~5.5V
Active Current (max.)	8mA	20mA	20mA	10mA	10mA	10mA	15mA
Operating Temperature	-40~+85°C	-10~+70°C	-10~+70°C	-40~+85°C	-40~+85°C	-40~+85°C	-40~+85°C
Packaging	44-QFP 42-DIP (1M ROM)	16-DIP CHIP	16-DIP CHIP	18-DIP 24-SOP	18-DIP 24-SOP CHIP	18-DIP 24-SOP CHIP	64-QFP
Notes	4-channels, echo	Single channel playback from OTPROM.	Single channel playback from OTPROM.	Speech peripheral w/o memory	Playback of PCM and OkiPCM.	Playback of PCM and OkiPCM.	8-Channels, Stereo, New OkiADPCM

• Typical output times are given for 4 KHz sampling frequency and with the chip´s own addressing capability without expansions.
• OkiPCM is a modified PCM for better sound performance, OkiADPCM2 is a further developed OkiADPCM with higher computation accuracy.

Specification Overview

Mask ROM Playback LSIs

PARAMETER	DATA	COMMON FEATURES
Algorithm	OkiADPCM/PCM	- Edit ROM Function
ADPCM Data Length	4-bit	- 2-Channel Playback
PCM Data Length	8-bit	- Echo Playback
Osc. Frequency (Hz)	256K/4096K	- Melody Generation
Sampling Rate (KHz)	4~32	- Beep Tone Output
Internal Memory (bit)		- Fade-in/out
MSM6652A	288K ROM (16.9sec)	- Parallel Control
MSM6653A	544K ROM (31.2sec)	- Serial Control
MSM6654A	1M ROM (63.8sec)	- OTP-Version 1MBit (MSM66P54, 20-DIP +3.5~5.5V/20mA max., -10~+70°C)
MSM6655A	1.5M ROM (96.5sec)	
MSM6656A	2M ROM (129.1sec)	
MSM6658A	4M ROM (260.0sec)	- ROM-less 64MBit (MSM6650, 64-QFP, 64-SDIP, +2.7~5.5V, 10mA max., -40~+85°C)
Number of Phrases	127	
Internal Low Pass Filter	-40dB/oct.	
Internal D/A Converter	12-bit	
Supply Voltage	+2.7~5.5V	
Active Current (max.)	10mA	
Operating Temperature	-40~+85°C	
Packaging	18-DIP* 24-SOP CHIP	

• Typical output times are given for 4 KHz sampling frequency in ADPCM mode.
* Package not for MSM6658A

Voice Registers

PARAMETER	MSM6389C	MSM63V89C	MSM6684A	MSM6685	MSM6595A	MSM6596A	MSM6597A
Function	Serial Reg	Serial Reg	Serial Reg	Serial Reg	Serial ROM	Serial ROM	Serial ROM
Organisation	1.048.576x1	1.048.576x1	4.194.304x1	8.388.608x1	1.048.576x1	1.048.576x2	1.048.576x3
Capacity (bit)	1M	1M	4M	8M	1M	2M	3M
Address Units	1024bits	1024bits	1024bits	1024bits	1024bits	1024bits	1024bits
Serial Access Time	3.0μs	3.0μs	1.5μs	1.5μs	1.5μs	1.5μs	1.5μs
Serial Read/Write Time	4.0μs	4.0μs	2.5μs	2.5μs	2.5μs	2.5μs	2.5μs
Operating Voltage	+3.5~5.5V	+2.7~3.6V	+3.5~5.5V	+3.5~5.5V	+3.0~5.5V	+3.0~5.5V	+3.0~5.5V
Max. Active Current	5mA	TBA	20mA	20mA	20mA	20mA	20mA
Max. Standby Current	100μA	TBA	150μA	200μA	10μA	10μA	10μA
Operating Temperature	0~+70°C	0~+70°C	0~+70°C	0~+70°C	-40~+85°C	-40~+85°C	-40~+85°C
Packaging	18-QFJ 16-DIP	18-QFJ	26-SOJ	26-SOJ	18-QFJ 24-SOP	18-QFJ 24-SOP	24-SOP

Serial voice ROMs must be used in conjunction with at leat one voice register or DRAM because the speech parameter header information is typically stored in volatile memory.

Special Purpose

MSM6295 - Choice For Musicians

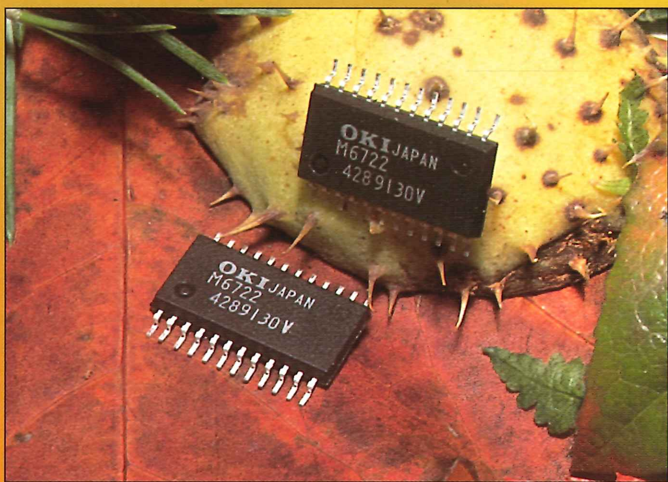
Profiting from the high audio quality of the OkiADPCM algorithm, this LSI manages the mixing of four reproduction channels with individual controlling facilities related to level attenuation and channel selection. There are two types which interface with common CPUs. MSM6295 addresses 2 Megabit external (EP)ROM, while MSM6295V is designed for 1 Megabit (EP)ROM. The memory contains both the sound/speech phrases and the corresponding addresses of the individual phrase locations in a header area. Analog portions shall be designed externally to permit maximum design flexibility. High sampling rates up to 32KHz would permit applications such as musical instruments, rhythm generators, BGM, echo generation, etc. In the pipeline is an 8-channel, stereo device, the MSM9810.

MSM6295: FOUR-CHANNEL MIXING SYNTHESIS

MSM6722 - Voice Changer

MSM6722 performs the pitch control of analog signals, i.e., voice. The pitch of an incoming signal can be shifted in real-time within the upper and lower octave relative to the base pitch level of the original input. What this device does in practice is to change the input voice to produce, for instance, a hoarse mickey mouse voice or similar. The input is sampled at 8KHz while the DAC varies the sampling rate between 4 and 16KHz in real-time. There are two different pin selectable control modes, one is the 17-stage up/down push button control, the other mode uses a four bit binary control interface.

Internally, the device comprises OpAmps, input low-pass filter, an 8-bit ADC, data processing unit (incl. 1 kilobit RAM), a 9-bit DAC and a buffered output low-pass filter, all of which represent the signal path from input to output. MSM6722 comes in a 24-pin small outline package (SOP) and can be used as voice changer in telephones to discourage unwanted callers.



MSM6722: NOT JUST A TOY

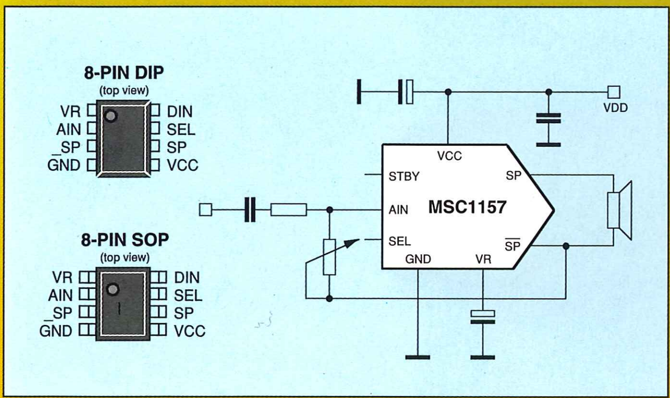
MSM9862 - Zero Retention With Flash

Currently under development, MSM9862 is a speech LSI for recording and playback particularly designed as core LSI for variable outgoing messages in answering machines. Because of its zero-retention feature, it is also an ideal component for voice memos or messaging in handy phones. On-chip it provides 512Kbit flash EEPROM and is also equipped with a DTMF receiver allowing for remote access to the answering machine (MCU controlled mode only). A serial interface provides direct connectability with OKI's serial voice registers and serial voice ROMs to expand recording capacity using an external means.

The built-in EEPROM allows a maximum recording time of 32 seconds, while sampling frequencies are selectable from 4, 5.3, 6.4 and 8.0KHz. Internally provided are a 12-bit ADC, 12-bit DAC, a low-pass filter and operational amplifiers with automatic gain control. The typical clock frequency is 4.096MHz.

MSM9862: FLASH EEPROM BUILT-IN

Amplifiers

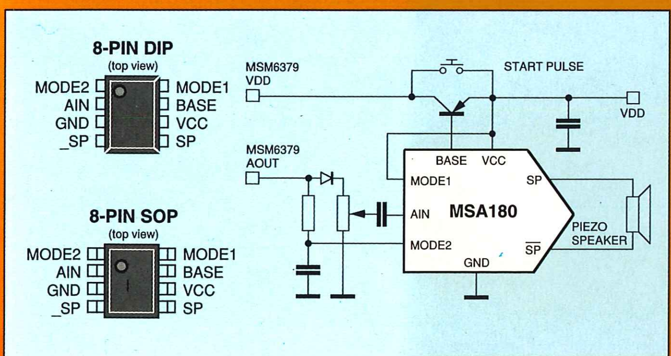


TYPICAL CONNECTION DIAGRAM FOR MSC1157

MSC1157 - Small Power Pack

MSC1157 has been developed for use with OKI´s synthesiser LSIs which can be operated at low supply voltages. This Bi-CMOS amplifier circuit essentially comprises two operational amplifiers, a bias circuit and a stand-by detector provision. A standby input is provided to simplify the matching with speech LSIs avoiding power loss when no voice signal is present. MSC1157 can be connected directly to a speaker without a coupling capacitor to achieve an audio power of 440mW typically into 32Ω load. Gain adjustment is possible by means of external resistors. A flexible power supply range from +2V to +6V and low current requirement of 1.56mA typically are essential characteristics for battery operation. The amplifier is available in chip form or packaged in a standard 8-pin DIP or an 8-pin SOP.

MSC1157 AND MSA180 COME IN 8-PIN DIP AND 8-PIN SOP



CONNECTION FOR MSA180 WITH MSM6379 SYNTHESISER

MSA180 - Drives a Piezo Directly

MSA180 is a piezo speaker driver amplifier for OKI Electric's range of speech LSIs. Its voltage gain can be adjusted to a factor of typically 10. The differential output provides an amplitude of twice the voltage supply. Two standby activation modes are built-in to simplify the matching with speech LSIs, avoiding power loss when no voice signal is present. Also provided is an output that can be connected to the base of an external transistor to control system supply current. A flexible power supply range from +2V to +6V and low current requirement (4.2mA typ.) are essential characteristics for battery operation. MSA180 is available in chip form or packaged in a standard 8-pin DIP or an 8-pin SOP.

The ADPCM Algorithm

ADPCM, Adaptive Differential Pulse Code Modulation, differs from other methods in that it does not truly synthesise speech. It initially involves digitising, compressing and storing actual analog sounds (digital recording). To re-create the original sound, the compressed digital word is expanded back to its original size, converted into an analog signal, amplified, and played through a speaker.

In fact, any kind of sound...voice, music, sound effects...etc., can be digitised, stored, and re-created faithfully with a high degree of naturalness. The idea behind this concept is to reduce the effective data rate in order to use memory efficiently and to minimise the data amount when transmitting voice data, while maintaining sufficient redundancy for faithful reproduction accuracy. In other words, maximum intelligibility and naturalness for all applications.

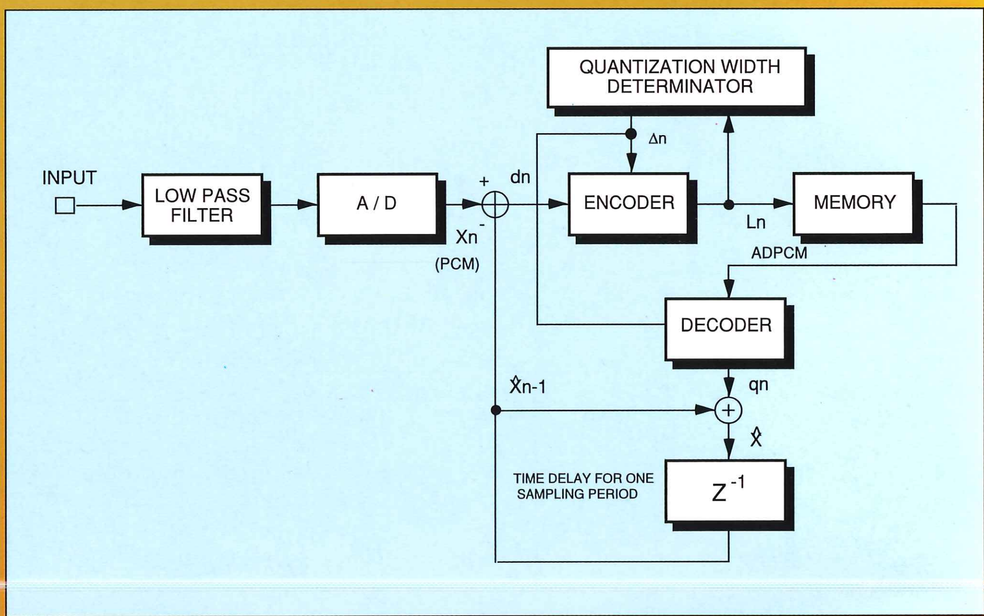
ADPCM represents an improvement with respect to conventional techniques in that it adaptively changes the quantiser step size (scale factor) to suit the waveform being encoded.

The drawing shows the block diagram of a typical ADPCM encoder. Input audio is filtered and then digitised by an A/D converter. This PCM input data, Xn, is compared with a PCM signal estimate, ^Xn-1, calculated from the previous sample Xn-1. The resulting differential value, dn, is encoded relative to the current step size, (Δn), and output as an ADPCM data sample, Ln. The ADPCM data is also fed to the step size determination logic and the decoder.

The ENCODER accepts the differential value, dn, from the comparator and the step size, Δn, and calculates a 4-bit ADPCM code. The DECODING logic transforms the output ADPCM data, Ln, back to a differential value, using the same step size data as the encoder. These differential values are accumulated to form an estimated (reproduced) waveform value, ^X. The value for the previous sample is fed back to the comparator as ^Xn-1. This feedback loop allows the system to recover from inputs that temporarily overload the encoder.

The DECODER accepts ADPCM code values, L, and step size values, Δ, calculates a differential value, q, and accumulates an estimated waveform value, ^X.

When the ADPCM device is reset, the step size, Δn, is set to minimum and the estimated waveform value, ^X, is set to zero (half scale). The system is almost identical to a typical DPCM coder except that additional logic provides adaption of quantiser step size on the basis of the most recent ADPCM code output.



Unlike true synthesisers, ADPCM is a sophisticated representative of waveform coders offering 4-bit accuracy. One bit, the MSB, is used as a sign bit, which denotes the direction of the original waveform, ascending or descending. Consequently, nearly any sound and any speech irrespective of the language can be processed and reproduced with surprising quality.

In order to achieve telephone bandwidth, 8 kHz sampling is the right choice, as the filter cut-off should be fixed on about(fs/2)*0.85.

For recorder devices, it is recommended that the input amplitude of the analog source signal be limited to about 80% of the dynamic range of the internal or external A/D converter.

It is recommended that the components for external output filters and amplifiers be carefully selected. An incorrect choice would impair the original quality the speech chips are designed to produce. This consideration equally includes the careful separation of analog and digital lines, the grounding of analog lines at both ends and further adequate separation from high speed digital circuits to avoid distortions thereof.

Maximum reproduction quality can be achieved with devices incorporating a 12-bit A/D converter and 12-bit D/A converter offering the fine resolution of 4096 vertical steps. An 8-bit device provides 256 steps, which makes a noticable difference in sound perception. The time for which a chip produces speech is determined by the available memory capacity and the bit-rate. The latter is composed of the product of the sampling frequency and the number of bits per sample (typically 4 bits).

Sample Routine for MSM80C154 Interfaced With MSM6388

A typical sample program shows the amount of programming effort for essential control tasks.

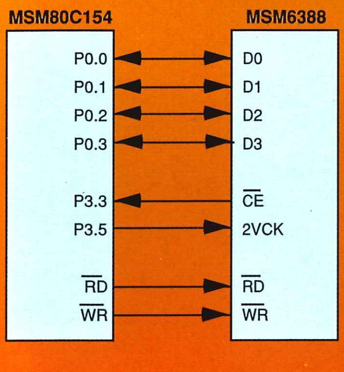
The assembler routine written for an Intel-compatible MSM80C154 demonstrates the functions of MSM6388. "EXEC" is a subroutine which sends a command to MSM6388 and sends or receives the corresponding data to/from the recorder chip. The complete command is passed to the subroutine in the internal RAM starting from the address "COMND" which must be bit addressable. The address COMND contains the MSM6388 command in the low order four bits. The high order nibble must contain all zeros. Successive addresses contain the corresponding data nibbles for certain commands in the low order four bits. The commands DTRD, DTWR, EREC and EPLAY are not supported by this example routine.

The command table, status register and hardware configuration are given for reference on this page. Most of the commands are one-nibble instructions which are executed upon transmission. SAMP selects the sampling frequency while CHAN determines one of 8 channels. Both commands expect another data nibble to follow. Address read/write operation requires maximum three subsequent more nibbles containing the relevant addresses.

More than 8 channels can be defined if the controller provides the start and stop addresses. In this case, channel 0 must be selected. Start and stop addresses are then written into the index area of channel 0 and fetched by MSM6388 prior to a recording or playback operation.

COMMAND	CODE	2. NIBBLE	3. NIBBLE	4. NIBBLE	FUNCTION
NOP	0000	-	-	-	NO OPERATION
INIT	0001	-	-	-	INITIALIZATION
PLAY	0010	-	-	-	SET PLAYBACK
REC	0011	-	-	-	SET RECORDING
START	0100	-	-	-	START FUNCTION
STOP	0101	-	-	-	STOP FUNCTION
SAMP	0110	- S2 S1 S0	-	-	SELECT SAMPLING FREQUENCY
CHAN	0111	- C2 C1 C0	-	-	SELECT CHANNEL
STWR	1000	A3 A2 A1 A0	A7 A6 A5 A4	A11 A10 A9 A8	WRITE A START ADDRESS
SPWR	1001	A3 A2 A1 A0	A7 A6 A5 A4	A11 A10 A9 A8	WRITE A STOP ADDRESS
STRD	1010	A3 A2 A1 A0	A7 A6 A5 A4	A11 A10 A9 A8	READ A START ADDRESS
SPRD	1011	A3 A2 A1 A0	A7 A6 A5 A4	A11 A10 A9 A8	WRITE A START ADDRESS
DTRD	1100	-	-	-	READ DATA (PLAYBACK MODE)
DTWR	1101	-	-	-	WRITE DATA (RECORD MODE)
EPLAY	1110	-	-	-	EXTERNAL PLAYBACK (SRAM/DRAM)
EREC	1111	-	-	-	EXTERNAL RECORD (SRAM/DRAM)

STATUS REGISTER	D3	D2	D1	D0
	"L"	DTMNT	RPMNT	BUSY



Though tailored for MSM6388, by knowledge of the instruction table of MSM6588, MSM6688 and MSM6789A this routine is easily adaptable. For details, reference is made to the latest Voice LSI data book.

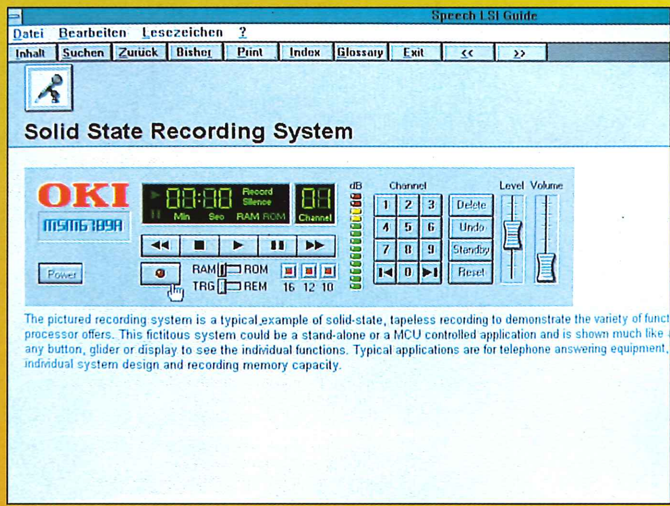
```
THIS ROUTINE PASSES A 6388-COMMAND TO THE SUBROUTINE EXEC
;
;R0=COMMAND DATA POINTER
;R1=COMMAND NIBBLE COUNTER FOR SAMP, CHAN, STWR, SPWR, STRD, SPRD
;ROUTINE DOES NOT SUPPORT COMMANDS DTRD, DTWR, EPLAY, EREC

GETD:     CE    EQU    P3.3            ;PORT 3.3 = CE 6388
          VCK   EQU    P3.5            ;PORT 3.5 = 2VCK 6388
        COMND   EQU    20H             ;ADDRESS IN INTERNAL RAM 80C154
          NOP   EQU    00000000B       ;COMMAND DATA
          INIT  EQU    00000001B
          PLAY  EQU    00000010B
          REC   EQU    00000011B
          START EQU    00000100B
          STOP  EQU    00000101B
          SAMP  EQU    00000110B
          CHAN  EQU    00000111B
          STWR  EQU    00001000B
          SPWR  EQU    00001001B
          STRD  EQU    00001010B
          SPRD  EQU    00001011B

                MOV    R0, #COMND
                MOV    @R0, #SPWR      ;EXAMPLE: SPWR
                INC    R0
                MOV    @R0, #A1        ;2. NIBBLE
                INC    R0
                MOV    @R0, #A2        ;3. NIBBLE
                INC    R0
                MOV    @R0, #A3        ;4. NIBBLE
                CLR    CE              ;ENABLE 6388, CE=0
                CALL   EXEC

EXEC      :     MOV    R0, #COMND      ;COMMAND INPUT
                MOV    A, @R0          ;GET COMMAND
                ADD    A, #0FAH        ;11111010 + COMMAND, COMMAND<6?
                MOV    R1, #1          ;DATA COUNTER
                JNC    AC_ST           ;SINGLE COMMAND

                INC    R1
                JNB    COMND.3, AC_ST  ;JUMP IF SAMP, CHAN
                MOV    R1, #4          ;4 NIBBLE COMMANDS

AC_ST     :     MOV    A, @R0          ;LOAD DATA FROM 6388 TO ACC
                JB     ACC.0, $-1      ;BUSY CHECK, LOOP UNTIL BUSY=0
                MOV    A, @R0          ;GET COMMAND
                MOVX   @R0, A          ;WRITE COMMAND
                INC    R0
                DJNZ   R1, AC_ST1
AC_STRT   :     MOV    A, COMND        ;GET COMMAND
                CJNE   A, #4, AC_STOP
                MOVX   A, @R0          ;GET BUSY
                JB     ACC. 0, $-1     ;BUSY CHECK, LOOP UNTIL BUSY=0
                SJMP   MA_RT           ;READY

AC_STOP   :     CJNE   A, #5, MA_RT    ;STOP COMMAND
                MOVX   A, @R0          ;GET BUSY FLAG
                JB     ACC.1, $-1      ;RPMNT CHECK
                SJMP   MA_RT           ;READY

AC_ST1    :     JB     COMND. 2, AC_ST ;JUMP IF SAMP & CHAN
                INC    R1

AC_AD     :     MOV    R2, #3          ;WAIT 3 VCK CYCLES
AC_AD1    :     JB     VCK, $          ;
                JNB    VCK, $          ;
                DJNZ   R2, AC_AD1
                DJNZ   R1, AC_AD2
                SJMP   MA_RT           ;READY

AC_AD2    :     JB     COMND. 1, AC_AD3 ;JUMP IF STRD, SPRD
                MOV    A, @R0
                MOVX   @R0, A          ;WRITE COMMAND
                SJMP   AC_RT
AC_AD3    :     MOVX   A, @R0          ;READ COMMAND
                MOV    @R0, A

AC_RT     :     INC    R0              ;DATA SET
                SJMP   AC_AD

MA_RT     :     SETB   CE              ;DISABLE 6388
                RET
```

Speech LSIs on Floppy

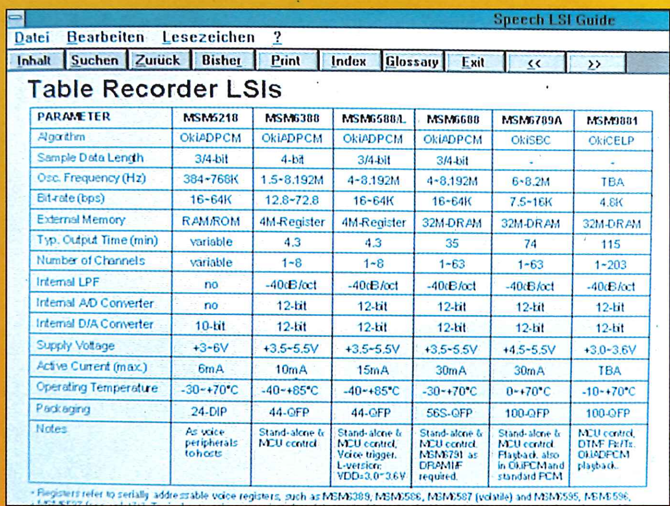
OKI Speech LSI data which is beyond the scope of this brochure is contained on a floppy for Windows ®PCs which is available now from OKI Electric Europe GmbH.

It is an electronic catalogue in the form of a Windows® Help file which allows you to browse through the topics and find desired information quickly and in many convenient ways. The electronic catalogue contains all speech LSIs and a few more in the pipeline. It has all pin assignments, many application diagrams, illustrations, and also includes topics explaining about speech algorithms. For quick reference, product line-up tables are provided. Further information deal with speech code development, the PC based development tool and a few products realised with OKI speech LSIs. An index topic allows quick search of items by alphanumerical order, while a glossary explains abbreviations and typical terms in conjunction with speech LSIs. Text hyperlinks allow quick jumps to glossary items. Most of the topics include scanned product photos.

The gratis catalogue is installed in the Windows® environment as any Help file is. Use a 32K or more colour display system to view images at their best. The software is supplied on as-is basis without warranty for any particular purpose. Further, it is not an integral part of a sales contract, since it does not substitute product data sheets. Updates will be provided upon request.



MSM6789A DEMONSTRATED BY MOUSE CLICKS ON BUTTONS



Speech LSI Guide

Datei Bearbeiten Lesezeichen ?

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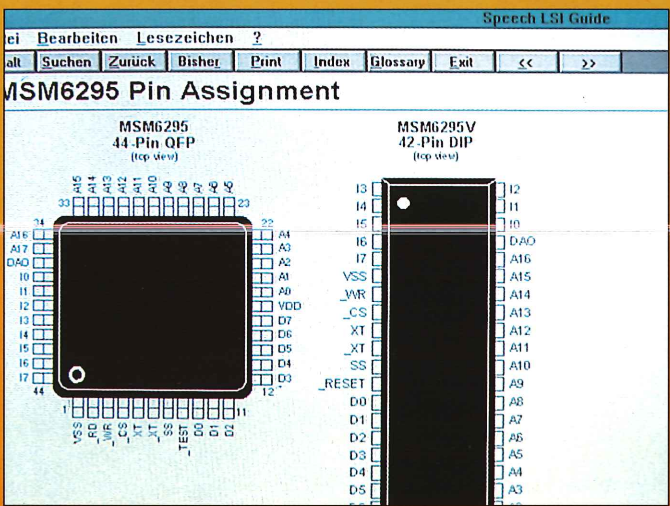
Table Recorder LSIs

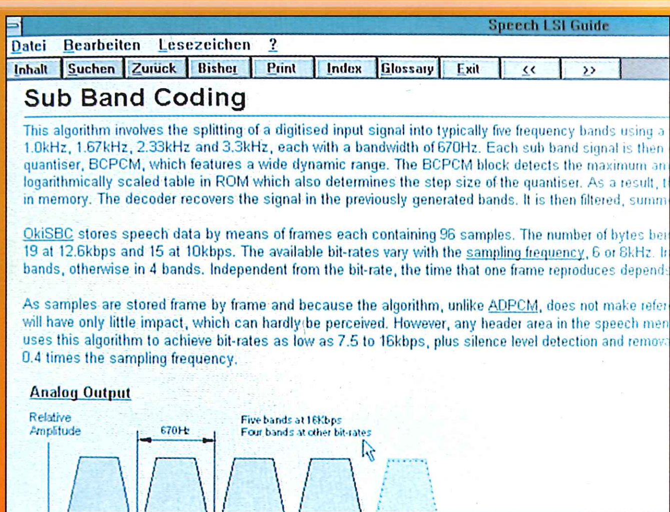
PARAMETER	MSM6218	MSM6388	MSM6588/L	MSM6688	MSM6789A	MSM9881
Algorithm	OkiADPCM	OkiADPCM	OkiADPCM	OkiADPCM	OkiSBC	OkiCELP
Sample Data Length	3/4-bit	4-bit	3/4-bit	3/4-bit	-	-
Osc. Frequency (Hz)	384~768K	1.5~8.192M	4~8.192M	4~8.192M	6~8.2M	TBA
Bit-rate (bps)	16~64K	12.8~72.8	16~64K	16~64K	7.5~16K	4.8K
External Memory	RAM/ROM	4M-Register	4M-Register	32M-DRAM	32M-DRAM	32M-DRAM
Typ. Output Time (min)	variable	4.3	4.3	35	74	115
Number of Channels	variable	1~8	1~8	1~63	1~63	1~203
Internal LPF	no	-40dB/oct	-40dB/oct	-40dB/oct	-40dB/oct	-40dB/oct
Internal A/D Converter	no	12-bit	12-bit	12-bit	12-bit	12-bit
Internal D/A Converter	10-bit	12-bit	12-bit	12-bit	12-bit	12-bit
Supply Voltage	+3~6V	+3.5~5.5V	+3.5~5.5V	+3.5~5.5V	+4.5~5.5V	+3.0~3.6V
Active Current (max.)	6mA	10mA	15mA	30mA	30mA	TBA
Operating Temperature	-30~+70°C	-40~+85°C	-40~+85°C	-30~+70°C	0~+70°C	-10~+70°C
Packaging	24-DIP	44-QFP	44-QFP	56S-QFP	100-QFP	100-QFP
Notes	As voice peripherals to hosts	Stand-alone & MCU control	Stand-alone & MCU control, Voice trigger. L-version: VDD=3.0~3.6V	Stand-alone & MCU control. MSM6791 as DRAM I/F required.	Stand-alone & MCU control Playback, also in OkiPCM and standard PCM	MCU control, DTMF Rx/Tx. OkiADPCM playback.

* Registers refer to serially addressable voice registers, such as MSM6389, MSM6586, MSM6587 (volatile) and MSM6595, MSM6596,

COMPREHENSIVE PRODUCT LINE-UP TABLES



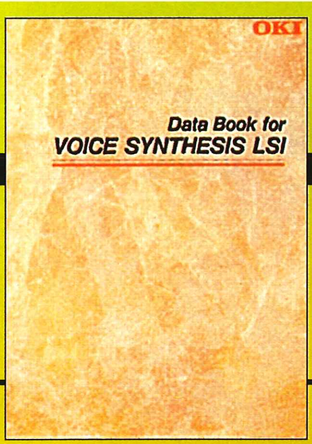
PIN ASSIGNMENTS ARE ALSO INCLUDED



PRINCIPLE OF SPEECH ALGORITHMS IN BRIEF

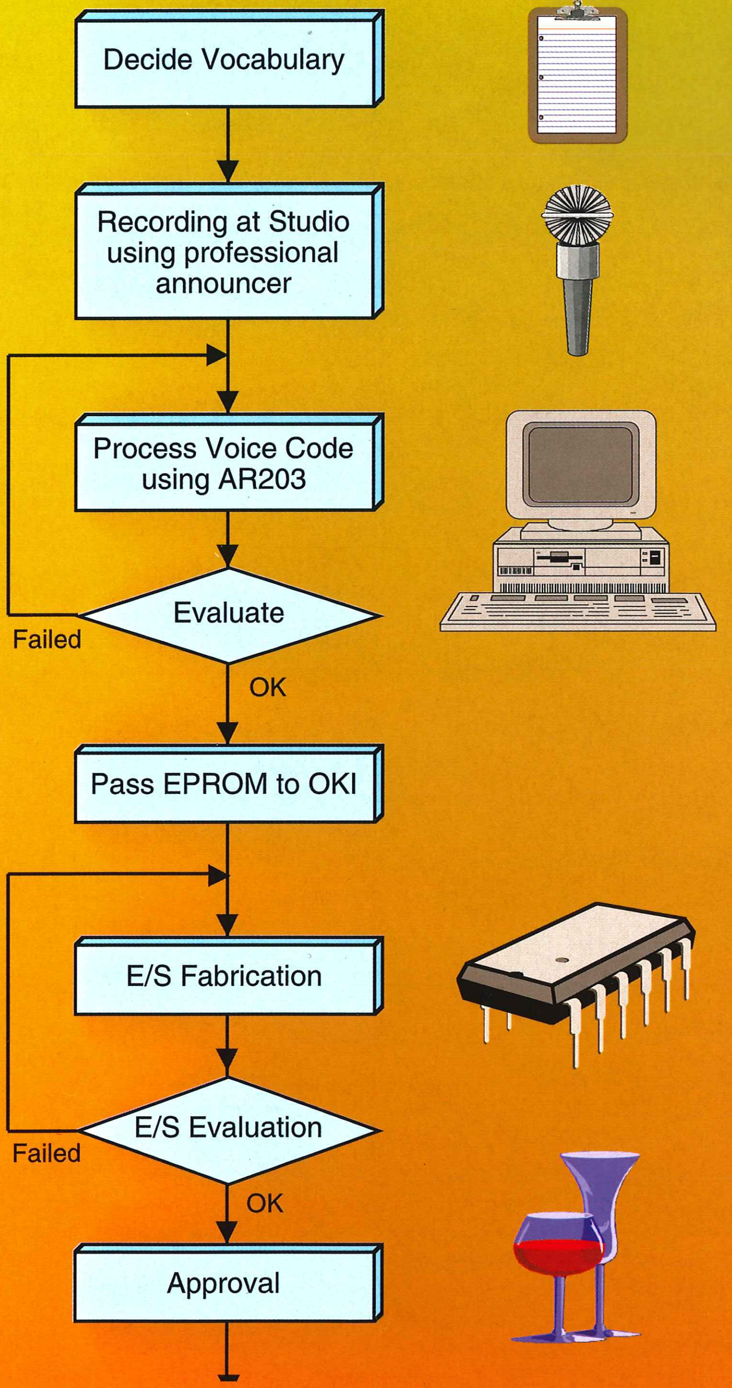
System Requirements

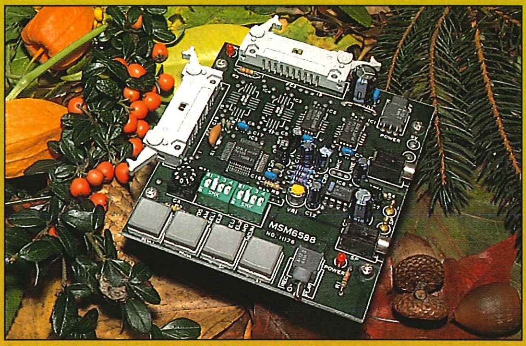
IBM PC 386/486/Pentium®, Windows® 3.1 or higher, mouse, 3 MBytes free RAM, hard disk, graphic card 256 colours min., better 32K or true colour.



Development Support

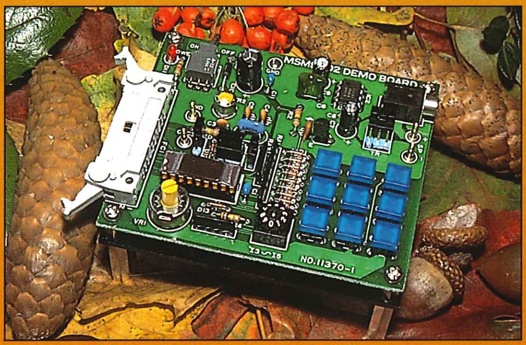
Typical Development Flow



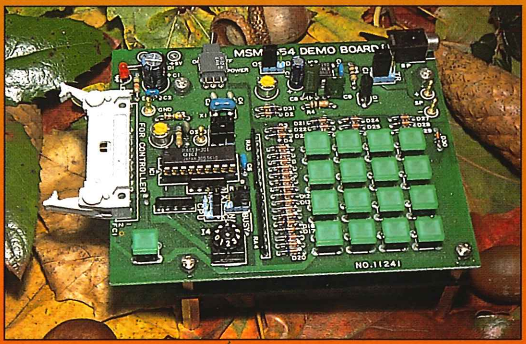


DEMO MODULE FOR MSM6588

DEMO MODULE FOR MSM6688

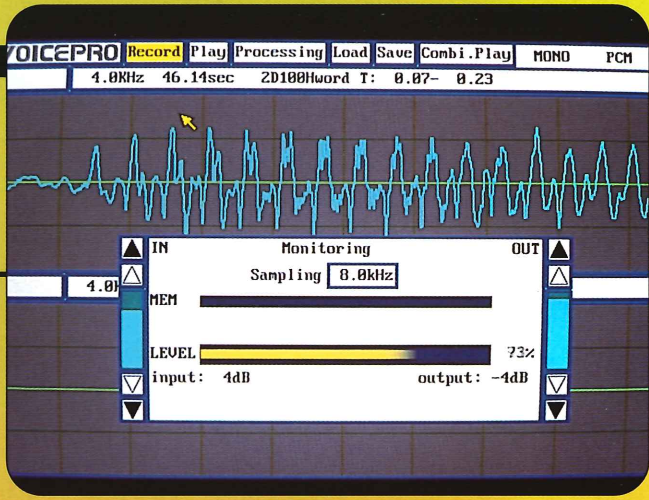


DEMO MODULE FOR MSM9802/MSM9803



DEMO MODULE FOR THE MSM6650 FAMILY

AR203



RECORDING SCREEN

In-House Speech Code Development

...eliminates the need for expensive analysis charges and gives you the freedom to edit your codes individually whenever and whereever required. Moreover, the ability to write EPROMs instantly shortens the code production process quite remarkably.

Outline of AR203

AR203 is fully PC supported. Included with the shipment is a standard 16-bit PC slot card, an EPROM programmer, a software driver and an English manual.

The host environment should consist of a PC-AT® or 100% compatible machine equipped with a harddisk, VGA® graphic adaptor and serial mouse. A printer is not essential. The driver software, VoicePRO runs under DOS 3.2 and higher on a 640K machine. However, for speech phrases of more than 16 seconds, EMS expansions are strongly recommended in order to record long phrases in one piece without concatenations. A tape deck or cassette player is needed as an analog source which can be connected directly to the installed board as well as a loudspeaker.

Hardware

The approximately 284(L) x 107(W) mm sized board represents an analog front end with 16-bit converters plus low pass filter featuring -48dB attenuation per octave. It further cores a compression engine receiving PCM data to provide ADPCM data for the target speech LSIs and vice-versa for the system´s playback function. Both analog and digital circuits are provided on board through to analog inputs and outputs for direct connection of a tape deck or microphone and a loudspeaker. Likewise, an external EPROM/OTP writer is linked to the card´s interface.

Optionally, a SBC board is offered to facilitate data compression for serial voice ROMs in conjunction with MSM6789A, the OkiSBC speech processor.

Performance

What AR203 does for you comprises all essentially demanded functions from recording all the way up to EPROM programming, plus a few extras. In detail:

- Sampling frequencies from 4 to 48kHz in 100Hz intervals
- Recording into host memory
  - Line input
  - Microphone
- Playback of a recording from host memory via speaker
- Stereo recording and playback, stereo speech file support
- Editing of a recorded voice file, including
  - Amplitude manipulation
  - Silencing / Silence insertion / Mixing
  - Fading / Cut / Copy / Paste and more
- Automatic pitch control
- HEX-file or BIN-file generation
- EPROM/OTP programming
- EPROM duplication
- Random access playback of speech files
- Covers all OKI speech LSIs, including OTP synthesisers
- Makes full use of Expanded Memory System, EMS

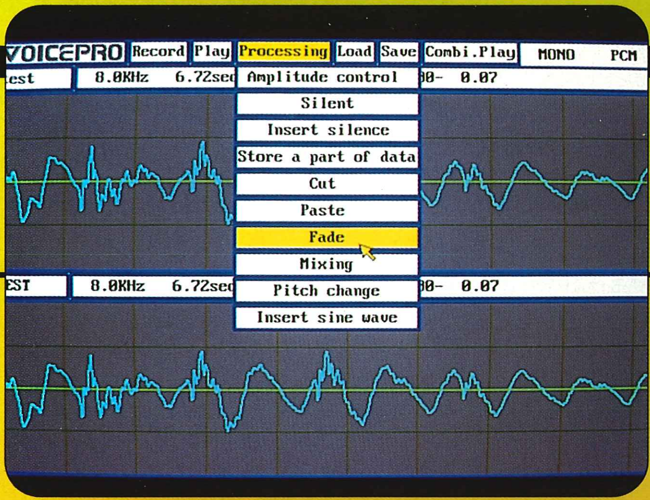
Software

The software driver, "VoicePRO", is fully graphics oriented and operable with keys or preferably with a mouse. The primary application of the software is for editing raw speech recordings. This includes amplitude amplification and attenuation, cutting, copying, insertion of selected excerpts and many more functions. Additionally, the auto-fade in/fade out function helps to save time. A recorded waveform is displayed on the graphic screen in several scalings allowing for precise editing at any point.

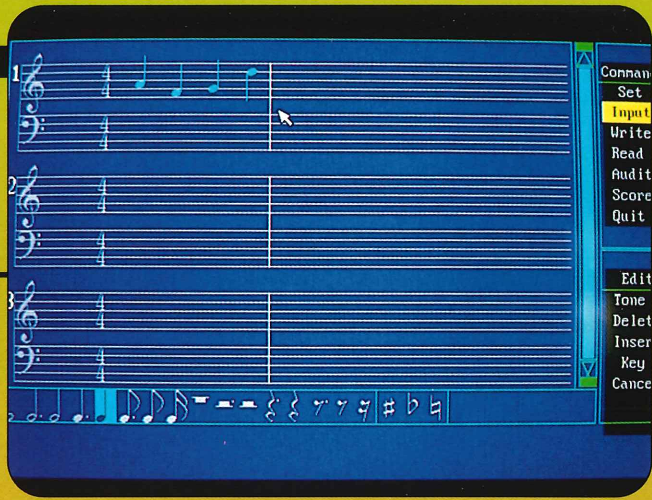
"VoicePRO" can be started up with a variety of option switches in order to adjust operating parameters for more convenience and easier further processing.

An optional software allows conversion from Windows® WAV files to PCM files which can be imported into VoicePRO and saved again to further convert to ADPCM data files.

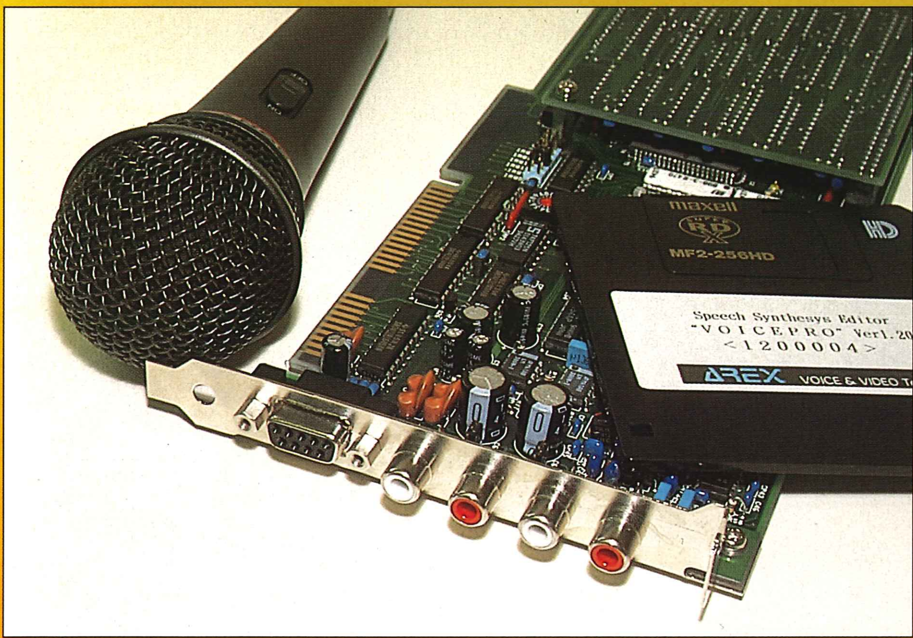
AR203, VoicePRO and PCM converter routine are products of AREX Co., Tokyo Japan, developed on behalf of OKI Electric Industry Co., Ltd.



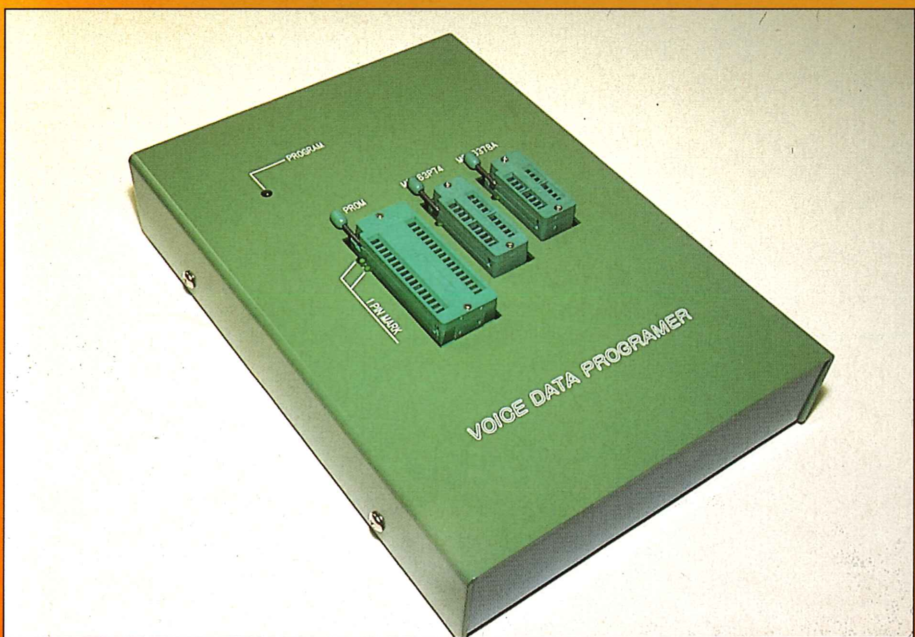
NUMEROUS EDITING FUNCTIONS



MELODY MAKER SCREEN



AR203 MAIN BOARD



AR203 EPROM/OTP PROGRAMMING ADAPTOR

KEY SPECIFICATION

- 16-bit A/D and D/A converter
- On-board low pass filter, -48dB/oct.
- Sampling frequencies from 4 to 48kHz
- Direct audio tape/microphone connection
- Direct speaker connection
- Line input impedance 10kΩ
- Line input level 1 Volt rms
- Speaker output power 0.5W/8Ω
- EPROM writing unit (included)
- Fully software controlled
- Writes CMOS EPROMs, 64k to 4096k
- Saves PCM/ADPCM and HEX files
- Mouse and key operable
- EMS capable up to theoretical 32 Megabytes
- Supports all OKI speech LSIs as targets
- Recording time: memory size/(f_{sample} x 20)

SYSTEM CONFIGURATION

Host Computer
IBM PC-AT®or 100% compatibles,
Upward compatible with 386 and 486.

Graphic Adaptor
VGA® is supported.

Operation System
MS-DOS® Ver 3.2 and higher.

Mouse
Microsoft® bus mouse or serial mouse.

Printer
Not essential.

Tape Deck
Cassette deck or open reel. With respect to higher audio quality an open reel tape is preferable.

Speaker
When using the built-in amplifier connect a 8Ω speaker designed for 5 Watts power.

For further information please contact:

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